THE UNITED STATES PATENT AND TRADEMARK OFFICE
BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

Applicant(s): Thomas Brumm et al.
Appl. No.: 09/827,487
Conf. No.: 5738
Filed: August 9, 2001
Title: SYSTEM FOR CONNECTING TELECOMMUNICATION EQUIPMENT TO A PACKET-SWITCHING TELECOMMUNICATION NETWORK
Art Unit: 2616
Examiner: Michael J. Moore, Jr.
Docket No.: 112740-207

Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

APPEAL BRIEF

Sir:

The Appellants submit this Appeal Brief in support of the Notice of Appeal filed on February 3, 2006. This Appeal is taken from the Final Rejection dated August 3, 2005, which is attached as Appendix B.

I. REAL PARTY IN INTEREST

The real party in interest for the above-identified patent application on appeal is Siemens Aktiengesellschaft by virtue of an Assignment recorded at the United States Patent and Trademark Office on August 9, 2001, at reel 012069, frames 0386-0390.

II. RELATED APPEALS AND INTERFERENCES

Appellants, Appellant's legal representative and the Assignee of the above-identified patent application do not know of any prior or pending appeals, interferences or judicial proceedings which may be related to, directly affect or be directly affected by or have a bearing on the Board's decision with respect to the above-identified Appeal.

III. STATUS OF CLAIMS

Claims 1 and 3-27 are pending in the above-identified patent application, with claims 1, 21, 24 and 26 being the only independent claims. Claims 1 and 3-27 stand rejected. Accordingly, claims 1 and 3-27 are being appealed in this Brief. A copy of the appealed claims is attached as Appendix A.

IV. STATUS OF AMENDMENTS

Claims 1, 21, 24 and 26 were amended in the previous Response to the final Office Action dated August 3, 2005, but only to correct minor informalities. No substantive amendments were made in the application after the final rejection.

V. SUMMARY OF CLAIMED SUBJECT MATTER

The present invention as recited in independent claims 1, 21, 24 and 26 is directed to a system and apparatuses for connecting a telecommunications device to a packet-switching communication network that includes the use of an interface unit. The interface unit is coupled to both the packet-switching communications network and to the telecommunication device. The telecommunications device is also further communicatively coupled to a line-switching telecommunications network.

In pertinent part, either the interface unit or a control unit coupled to the interface unit are implemented for converting at least some of the first signaling data of the packet-switching communications network into a second signaling data of the line-switching communications network. In this case, the first signaling data is used for packet-switching communications, while the second signaling information is used for line-switching communications. The converted second signaling data is then fed to the telecommunication device. This conversion technique can also be used for communicating signaling data from the telecommunication devices to an IP network. In other word, this conversion technique can involve the conversion of H.323 signaling data into DSS1 signaling data and vice versa.

An important aspect of the invention is that the second signaling data [*of the line-switching communication network*] is transmitted to the packet-switching communications network <u>instead</u> of the first signaling data [*of the packet-switched communication network*] when the second signaling data cannot be converted to the first signaling data (see, Applicant's Application, page 14, line 17-page 15, line 15).

Although specification citations are given in accordance with C.F.R. 1.192(c), these reference numerals and citations are merely examples of where support may be found in the specification for the terms used in this section of the Brief. There is no intention to suggest in any way that the terms of the claims are limited to the examples in the specification. Pointing out specification support for the claim terminology as is done here to comply with rule 1.192(c) does not in any way limit the scope of the claims to those examples from which they find support. Nor does this exercise provide a mechanism for circumventing the law precluding reading limitations into the claims from the specification. In short, the references numerals and

specification citations are not to be construed as claim limitations or in any way used to limit the scope of the claims.

VI. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

Claims 1 and 3-27 stand rejected under 35 U.S.C. §103(a) as being unpatentable over *Rose* et al. (US Patent 6,396,840) in view of *Ress et al.* (US Patent 6,885,658).

VII. ARGUMENT

A. LEGAL STANDARDS

In making a determination that an invention is obvious, the Patent Office has the initial burden of establishing a *prima facie* case of obviousness. *In re Rijckaert*, 9 F.3d 1531, 1532, 28 U.S. P.Q.2d 1955, 1956 (Fed. Cir. 1993). "If the examination at the initial stage does not produce a *prima facie* case of unpatentability, then without more the applicant is entitled to grant of the patent." *In re Oetiker*, 24 U.S.P.Q.2d 1443, 1444 (Fed. Cir. 1992).

In order to maintain a *prima facie* case of obviousness under 35 U.S.C. §103, the Examiner must satisfy the following criteria: 1) a suggestion or motivation, either in the cited references themselves or in the knowledge generally available to one of ordinary skill in the art, to modify the references or combine their teachings to arrive at the invention; 2) a reasonable expectation of success at arriving at the invention, if the combination of the cited references is made; and 3) a teaching of suggestion of all the recited claim limitations in the combination of the cited references. (see MPEP 2142).

The teaching or suggestion to make the claimed combination and the reasonable expectation of success must both be found in the prior art, and not based on applicant's disclosure. *In re Vaeck*, 947 F.2d 488, 20 USPQ2d 1438 (Fed. Cir. 1991). The initial burden is on the examiner to provide some suggestion of the desirability of doing what the inventor has done. "To support the conclusion that the claimed invention is directed to obvious subject matter, either the references must expressly or impliedly suggest the claimed invention or the examiner must present a convincing line of reasoning as to why the artisan would have found the claimed invention to have been obvious in light of the teachings of the references." *Ex parte Clapp*, 227 USPQ 972, 973 (Bd. Pat. App. & Inter. 1985). When the motivation to combine the teachings of the references is not immediately apparent, it is the duty of the examiner to explain why the combination of the teachings is proper. *Ex parte Skinner*, 2 USPQ2d 1788 (Bd. Pat. App. & Inter. 1986). (see MPEP 2142).

Further, the Federal Circuit has held that it is "impermissible to use the claimed invention as an instruction manual or 'template' to piece together the teachings of the prior art so that the claimed invention is rendered obvious." *In re Fritch*, 23 U.S.P.Q.2d 1780, 1784 (Fed. Cir. 1992). "One cannot use hindsight reconstruction to pick and choose among isolated disclosures in the prior art to deprecate the claimed invention" *In re Fine*, 837 F.2d 1071 (Fed. Cir. 1988).

Moreover, the Federal Circuit has held that "obvious to try" is not the proper standard under 35 U.S.C. §103. *Ex parte Goldgaber*, 41 U.S.P.Q.2d 1172, 1177 (Fed. Cir. 1996). "An-obvious-to-try situation exists when a general disclosure may pique the scientist curiosity, such that further investigation might be done as a result of the disclosure, but the disclosure itself does not contain a sufficient teaching of how to obtain the desired result, or that the claim result would be obtained if certain directions were pursued." *In re Eli Lilly and Co.*, 14 U.S.P.Q.2d 1741, 1743 (Fed. Cir. 1990).

B. THE REJECTION UNDER 35 U.S.C. §103(A) IS IMPROPER BECAUSE *ROSE ET AL.* IN VIEW OF *RESS ET AL.* DOES NOT RENDER OBVIOUS THE CLAIMED INVENTION

Claims 1 and 3-27 stand rejected under 35 U.S.C. §103(a) as being unpatentable over *Rose* et al. (US Patent 6,396,840) in view of *Ress et al.* (US Patent 6,885,658).

The cited art, alone or in combination, fails to disclose a system or apparatuses for processing first and second signaling data in a communications system, which is coupled to both packet-switched and line-switched communications network, "wherein the second signaling data [*of the line-switching communication network*] is transmitted in the packet-switching communications network instead of the first signaling data [*of the packet-switched communication network*] when the second signaling data cannot be converted to the first signaling data." The above features of the present invention are recited in independent claim 1, and similarly recited in independent claims 21, 24 and 26.

In the final Office Action, the Examiner conceded that *Rose* fails to teach the aforementioned limitations (see, Office Action, page 5). However, in the Office Action the Examiner relies on *Ress* for teaching or suggesting the above limitations, and for formulating the obviousness rejection. The Applicants respectfully disagree with the Examiner's interpretation of *Ress* and instead suggest that *Ress* also falls short of the present invention.

Ress discloses interworking agents that communicate with each other according to a protocol independent format referred to as the "agent interworking protocol" (AIP). The agent interworking protocol represents a superset of the messaging capabilities of all protocols to be supported within the packet network (see, *Ress*, col. 6, lines 21-37). The agent networking protocol is used by the system of *Ress* to transfer message data (SIP, MGCP, H.323), and not signaling information or data, according to a mapped protocol. In col. 9, lines 2-30, *Ress* discloses that the internetworking relates to the protocol of the message (H.323) that is tunneled (i.e., not converted), and not the signaling (H.245). The agent of *Ress* checks to see if the specific signaling is available, and if so, transfers the message data in a converted or native format. However, if the signaling is not supported, the data is simply discarded (see, *Ress,* col. 10, lines 35-41).

In the Advisory Action, the sections of *Ress* relied upon by the Examiner do not teach or suggest the optional (i.e., second instead of the first) sending of signaling data when line switched signaling data (second signaling data) cannot be converted to packet-switched data (first signaling data). Although *Ress* generally discloses signaling between the agents and a MGCP gateway (see, *Ress,* col. 11, lines 59-67), the entire disclosure of *Ress* is premised on the exchange of IP telephony protocols. In fact, line-switched signals are not even addressed. Furthermore, the H.245 signaling at col. 9, lines 17-30 merely teaches that message tunneling using H.245 signaling is accomplished by sending "H.245 indications" between two H.323 devices, where the H.245 signaling indicates terminal capabilities of the endpoint, and the agent acknowledges the message (i.e., the message goes through, see, *Ress*, col. 12, line 57–col. 13, line 6). The H.245 terminal capability messages and the AIP CPG messages are then combined into a multipart message, where if one or the other message protocols are not supported, they will be discarded (see, *Ress*, col. 13, lines 1-6). This cited section in *Ress* cannot be interpreted as meaning that the H.245 message is being <u>converted</u> to AIP CPG messages to establish optional transmission. Furthermore, as discussed previously, both the H.245 message and AIP CPG messages relied on in the embodiments cited by the examiner are transmitting in the packet-switched network (see, *Ress*, col. 6, lines 44-51).

To summarize, the *Ress* reference contemplates the use of a protocol-independent agent interworking protocol to correlate disparate protocols being used across a gateway to ultimately provide a protocol-neutral system. Additionally, *Rose* is specifically directed to routing calls between narrow-band and broad-band ISDN, and further provides already-translated messaging and signaling (see, *Rose*, col. 8, lines 22-36; col. 9, lines 6-22). There is simply no teaching, suggestion or motivation for one having ordinary skill in the art to rely on the teaching in *Ress* when considering the teaching of *Rose.* In fact, there is no provision in *Rose* whatsoever that would suggest the use of a protocol-neutral configuration or AIP messaging such as that taught in *Ress*, without relying on impermissible hindsight. Moreover, it is not understood how a configuration such as that in Ress would even be incorporated into the teaching of Rose.

In light of the above, the Applicants respectfully submit that Examiner has not satisfied the criteria necessary for maintaining a *prima facie* case of obviousness and independent claims 1, 21, 24 and 26 are patentable over the cited prior art.

VIII. CONCLUSION

Appellants respectfully submit that claims 1 and 3-27 are non-obvious in view of the cited art. Accordingly, the Appellants respectfully submit that the rejections of pending claims 1 and 3-27 are erroneous in law and fact and should be reversed by this Board.

Respectfully submitted,

BELL, BOYD & LLOYD LLC

BY ______________________

Peter Zura
Reg. No. 48,196
Customer No.: 29177
Phone: (312) 807-4208

Dated: April 21, 2006

APPENDIX A

APPENDIX A

PENDING CLAIMS ON APPEAL OF U.S. PATENT APPLICATION SERIAL NO. 09/646,442

Listing of Claims:

Claim 1. (previously presented): A system for connecting a telecommunications device to a packet-switching communications network, the system comprising:

at least one telecommunications device communicatively coupled to a line-switching communications network;

a packet-switching communications network, wherein first signaling data is transmitted between a first subscriber line and a second subscriber line of the packet-switching communications network; and

an interface unit connected to both the packet-switching communications network and the telecommunications device, the interface unit converting at least some of the first signaling data, which is intended for the subscriber line using the packet-switching communications network, into second signaling data of the line-switching communications network, and feeding the second signaling data to the telecommunications device, and vice versa,

wherein the second signaling data is transmitted to the packet-switching communications network instead of the first signaling data when the second signaling data cannot be converted to the first signaling data.

Claim 2. (canceled).

Claim 3. (previously presented). A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the first and second signaling data contain signaling messages.

Claim 4. (previously presented): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 3, wherein the interface unit, via an interface program, converts the signaling messages of the packet-switching

communications network into equivalent signaling messages of the line-switching communications network.

Claim 5. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 4, wherein the conversion is carried out using equivalent signaling messages stored in a database.

Claim 6. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 4, wherein the interface program for signaling messages to which no equivalent signaling message is assigned is transmitted using a data packet as user data.

Claim 7. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 4, wherein the interface program generates messages which at least one of the packet-switching communications network and the line-switching communications network requires as an acknowledgement of transmitted signaling data.

Claim 8. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 3, wherein the signaling messages are used to make connection setups between the first and second subscribers.

Claim 9. (previously presented): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 3 wherein the signaling messages are used for at least one of activating, deactivating, and registering at least one service feature.

Claim 10. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 9, wherein the service feature comprises at least one of call pick-up, three-way conferencing, large-scale conferencing, holding, displaying of toll information, a closed user group, call number identification, automatic callback

when busy, automatic callback when no response, call barring, an indication of call waiting and call transfer.

Claim 11. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 3, wherein the signaling messages are transmitted in the packet-switching communications network independently of user connections.

Claim 12. (previously presented): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 3, wherein the signaling message of the line-switching communications network are DSS1 messages.

Claim 13. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 3, wherein the signaling messages of the packet-switching communications network are signaling messages of the H.225 signaling protocol Standard.

Claim 14. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the telecommunications device is at least one of an ISDN telephone, an analog telephone, an analog modem, an ISDN modem and an analog facsimile device.

Claim 15. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the telecommunications device is a private branch exchange.

Claim 16. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the interface unit is arranged in a separate physical unit.

Claim 17. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the interface unit is a module in the telecommunications unit.

Claim 18. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein a control unit of the interface unit automatically logs on the interface unit as a subscriber to the packet-switching communications network.

Claim 19. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the interface unit has a control unit which converts the data using at least one program module.

Claim 20. (original): A system for connecting a telecommunications device to a packet-switching communications network as claimed in claim 1, wherein the packet-switching communications network is a network based on an Internet protocol.

Claim 21. (previously presented): An interface unit which is communicatively coupled to both a packet-switching communications network and to a telecommunications device that is further communicatively coupled to a line-switching telecommunications network, comprising:

a control unit which converts at least one item of signaling information of the packet-switching communications network into a second item of signaling information of a line-switching communications network and feeds it to the telecommunications device, and vice versa,

wherein the second item of signaling information is transmitted to the packet-switching communications network instead of the first item of signaling information when the second item of signaling information cannot be converted to the first item.

Claim 22. (original): An interface unit as claimed in claim 21, wherein the interface unit is used to connect a communications terminal to the packet-switching communications network.

Claim 23. (original): An interface unit as claimed in claim 21, wherein the interface unit is used to connect a private branch exchange to the packet-switching communications network.

Claim 24. (previously presented): A communications terminal, connected to a line-switching communications network and which is used for telecommunications, comprising:

an interface unit, communicatively coupled to a packet-switching communications network and to a telecommunications device, wherein said telecommunications device is further communicatively coupled to a line-switching telecommunication network;

a control unit, communicatively coupled to said interface unit that converts at least a part of a first item of signaling information of the packet-switching communications network into a second item of signaling information of a line-switching communications network and feeds it to the telecommunications device, and vice versa,

wherein the second item of signaling information is transmitted to the packet-switching communications network instead of the at least a part of the first item of signaling information when the second item of signaling information cannot be converted to the first item.

Claim 25. (original): A communications terminal as claimed in claim 24, wherein the interface unit is a module of the communications terminal.

Claim 26. (previously presented): A private branch exchange, connected to a line-switching communications network and which is used for telecommunications, comprising:

an interface unit which is communicatively coupled to both to a packet-switching communications network and to a telecommunications device, wherein said telecommunications device is further communicatively coupled to a line-switching telecommunications network;

a control unit, communicatively coupled to said interface unit that converts at least a part of a first item of signaling information of the packet-switching communications network into a

second item of signaling information of a line-switching communications network and feeds it to the telecommunications device, and vice versa, wherein the interface unit is used to connect the private branch exchange to the packet-switching communications network,

wherein the second item of signaling information is transmitted to the packet-switching communications network instead of the at least a part of first item of signaling information when the second item of signaling information cannot be converted to the first item.

Claim 27. (original): A private branch exchange as claimed in claim 26, wherein the interface unit is a module of the private branch exchange.

APPENDIX B

Final Office Action Mailed on August 3, 2005

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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/827,487	08/09/2001	Thomas Brumm	112740-207	5738

29177 7590 08/03/2005

BELL, BOYD & LLOYD, LLC
P. O. BOX 1135
CHICAGO, IL 60690-1135

EXAMINER	
MOORE JR, MICHAEL J	
ART UNIT	**PAPER NUMBER**
2666	

DATE MAILED: 08/03/2005

Due: 11-3-05

References Downloaded

Please find below and/or attached an Office communication concerning this application or proceeding.

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Office Action Summary	Application No.	Applicant(s)	
	09/827,487	BRUMM ET AL.	
	Examiner	Art Unit	
	Michael J. Moore, Jr.	2666	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *16 May 2005*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1 and 3-27* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1 and 3-27* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *16 May 2005* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☒ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application (PTO-152)

6) ☐ Other: _____.

DETAILED ACTION

Information Disclosure Statement

1. The information disclosure statement (IDS) submitted on 9/9/2002 is in compliance with the provisions of 37 CFR 1.97. Accordingly, the examiner has considered the information disclosure statement.

Drawings

2. Replacement drawings were received on 5/16/2005. These drawings are acceptable and have been entered.

Claim Objections

3. Claims **1, 21, 24, and 26** are objected to because of the following informalities:

Regarding claim **1**, on line 8, the word "to" after the word "both" is not needed. Also, on line 9, the word "to" after the word "and" is not needed. Also, on line 13, there is some confusion regarding the phrase "the second data". It is believed that this phrase should be "the second signaling data". Also, on line 13, the word "in" after the word "transmitted" should be "to". Lastly, on lines 14-15, there is some confusion regarding the phrase "the first data". It is believed that this phrase should be "the first signaling data".

Regarding claim **21**, on lines 8-9, there is some confusion regarding the phrase "the second item of signaling information". It is unclear which "item of signaling information" is being referred to. Also, on line 8, the word "in" after the word "transmitted" should be "to". Lastly, on lines 9-10, there is some confusion

regarding the phrase "the first item of signaling information". It is unclear which "item of signaling information" is being referred to.

Regarding claim **24**, on line 1, the phrase "can be connected" should be changed to "is connected" in order to constitute a positive limitation. Also, on line 13, the word "in" after the word "transmitted" should be "to". Lastly, on lines 14-15, there is some confusion regarding the phrase "the first item of signaling information". It is believed that this phrase should be "the part of the first item of signaling information".

Regarding claim **26**, on line 1, the phrase "can be connected" should be changed to "is connected" in order to constitute a positive limitation. Also, on line 13, the word "in" after the word "transmitted" should be "to". Lastly, on lines 14-15, there is some confusion regarding the phrase "the first item of signaling information". It is believed that this phrase should be "the part of the first item of signaling information".

Appropriate correction is required.

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

5. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that

the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of 35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

6. Claims **1 and 3-27** are rejected under 35 U.S.C. 103(a) as being unpatentable over Rose et al. (U.S. 6,396,840) ("Rose") in view of Ress et al. (U.S. 6,885,658) ("Ress").

Regarding claim **1**, Rose teaches an integrated system architecture in Figure 5 connecting subscriber terminal 119 (telecommunications device) to LAN 10 (packet-switching network). Rose also teaches subscriber terminal 119 (telecommunications device) that is connected to exchange 118 (line-switching network) as shown in Figure 5.

Rose also teaches LAN 10 (packet-switching communications network) of Figure 5 that communicates with multi-media endpoint 110 (second subscriber line). Rose also teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and subscriber terminal 119 (telecommunications device).

Rose also teaches gateway interface 112 of Figure 6 that translates H.225 call signaling (first signaling data) from LAN 10 into DSS1 broadband format

(second signaling data) for onward routing as spoken of on column 8, lines 53-65.

Rose fails to teach where the second signaling data is transmitted to the packet-switching communications network instead of the first signaling data when the second signaling data cannot be converted to the first signaling data.

However, Ress teaches a method of protocol interworking where message tunneling is used to transfer a native protocol message (second signaling data) from one protocol agent to another protocol agent without converting to and from the agent interworking protocol (first signaling) in the case that the native protocol message does not map to the other agent protocol as spoken of on column 9, lines 6-16.

At the time of the invention, it would have been obvious to someone skilled in the art to combine the tunneling teachings of Ress with the interworking teachings of Rose in order to communicate messages or parameters which do not map to any other agent protocols, but provide added value for a call between two devices as spoken of on column 9, lines 6-16 of Ress.

Regarding claim **3**, Rose further teaches H.225 RAS 22, H.225 call signaling 14, and H.245 negotiation control 26 (first signaling data) as well as call signaling 114 (second signaling data) shown in Figure 6 and spoken of on column 8, lines 53-59.

Regarding claim **4**, Rose further teaches gateway interface 112 (interface unit) of Figure 6 that translates incoming H.225 call signaling (signaling

messages) from LAN 10 (packet network) into DSS1 broadband format (signaling messages) for onward routing as spoken of on column 8, lines 53-65.

Regarding claim **5**, Rose further teaches memory 154 of gateway interface 112 of Figure 6 that contains look-up tables (database) associated with signaling protocol translation schemes used to translate LAN signaling to narrowband/broadband signaling as spoken of on column 8, line 66 – column 9, line 5.

Regarding claim **6**, Rose further teaches gateway interface 112 of Figure 6 that translates incoming H.225 call signaling (signaling messages) from LAN 10 (packet network) into DSS1 broadband format (signaling messages) for onward routing as spoken of on column 8, lines 53-65.

Regarding claim **7**, Rose further teaches a message (acknowledgement) confirming the trunk circuit identity sent from next exchange 118 to call handler 116 in response to a setup signaling message sent from call handler 116 to next exchange 118 as spoken of on column 10, lines 24-36.

Regarding claim **8**, Rose further teaches call signaling messages 114 that are used to set-up and clear-down calls as spoken of on column 7, lines 53-56.

Regarding claim **9**, Rose further teaches the H.225 RAS (registering, admission, and status) signaling shown in Figure 6.

Regarding claim **10**, Rose further teaches call signaling information 114 containing an address of a called party (call number identification) as spoken of on column 9, lines 13-18.

Regarding claim **11**, Rose further teaches gateway interface 112 of Figure 6 that translates incoming H.225 call signaling (signaling messages) from LAN 10 (packet network) into DSS1 broadband format (signaling messages) for onward routing as spoken of on column 8, lines 53-65.

Regarding claim **12**, Rose further teaches the DSS1 signaling format spoken of on column 8, lines 53-59.

Regarding claim **13**, Rose further teaches the H.225 RAS 22 and H.225.0 call signaling 14 spoken of on column 8, lines 44-49.

Regarding claim **14**, Rose further teaches subscriber terminal 119 of Figure 5 that utilizes ISDN broadband communication as spoken of on column 7, lines 50-62.

Regarding claim **15**, Rose further teaches the exchange 118 shown in Figure 5.

Regarding claim **16**, Rose further teaches gateway interface 112 shown in Figure 5.

Regarding claim **17**, Rose further teaches gateway interface 112 shown in Figure 5.

Regarding claim **18**, Rose further teaches the gateway interface 112 operating as a subscriber as spoken of on column 12, lines 11-16.

Regarding claim **19**, Rose further teaches gateway interface 112 shown in Figure 5.

Regarding claim **20**, Rose further teaches the H.225 RAS 22 and H.225.0 call signaling 14 spoken of on column 8, lines 44-49.

Regarding claim **21**, Rose teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and subscriber terminal 119 (telecommunications device) that is further connected to exchange 118 (line-switching network) as shown in Figure 5.

Rose also teaches processor 152 (control unit) of gateway interface 112 of Figure 6 that translates incoming H.225 call signaling (signaling information) from LAN 10 (packet network) into DSS1 broadband format (signaling information) for onward routing as spoken of on column 8, lines 53-65.

Rose fails to teach where the second signaling data is transmitted to the packet-switching communications network instead of the first signaling data when the second signaling data cannot be converted to the first signaling data.

However, Ress teaches a method of protocol interworking where message tunneling is used to transfer a native protocol message (second signaling data) from one protocol agent to another protocol agent without converting to and from the agent interworking protocol (first signaling) in the case that the native protocol message does not map to the other agent protocol as spoken of on column 9, lines 6-16.

At the time of the invention, it would have been obvious to someone skilled in the art to combine the tunneling teachings of Ress with the interworking teachings of Rose in order to communicate messages or parameters which do not map to any other agent protocols, but provide added value for a call between two devices as spoken of on column 9, lines 6-16 of Ress.

Regarding claim **22**, Rose further teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and subscriber terminal 119 (terminal).

Regarding claim **23**, Rose further teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and exchange 118.

Regarding claim **24**, Rose teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and subscriber terminal 119 (telecommunications device) that is further connected to exchange 118 (line-switching network) as shown in Figure 5.

Rose also teaches processor 152 (control unit) of gateway interface 112 of Figure 6 that translates incoming H.225 call signaling (signaling information) from LAN 10 (packet network) into DSS1 broadband format (signaling information) for onward routing as spoken of on column 8, lines 53-65.

Rose fails to teach where the second signaling data is transmitted to the packet-switching communications network instead of the first signaling data when the second signaling data cannot be converted to the first signaling data.

However, Ress teaches a method of protocol interworking where message tunneling is used to transfer a native protocol message (second signaling data) from one protocol agent to another protocol agent without converting to and from the agent interworking protocol (first signaling) in the case that the native protocol message does not map to the other agent protocol as spoken of on column 9, lines 6-16.

At the time of the invention, it would have been obvious to someone skilled in the art to combine the tunneling teachings of Ress with the interworking teachings of Rose in order to communicate messages or parameters which do not map to any other agent protocols, but provide added value for a call between two devices as spoken of on column 9, lines 6-16 of Ress.

Regarding claim **25**, Rose further teaches gateway interface 112 shown in Figure 5.

Regarding claim **26**, Rose teaches exchange 142 (private branch exchange) of Figure 5 connected to exchange 118 (line-switching network).

Rose also teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and subscriber terminal 119 (telecommunications device) that is further connected to exchange 118 (line-switching network) as shown in Figure 5.

Rose also teaches processor 152 (control unit) of gateway interface 112 of Figure 6 that translates incoming H.225 call signaling (signaling information) from LAN 10 (packet network) into DSS1 broadband format (signaling information) for onward routing as spoken of on column 8, lines 53-65.

Rose also teaches gateway interface 112 (interface unit) of Figure 5 connected to both LAN 10 (packet-switching network) and exchange 118.

Rose fails to teach where the second signaling data is transmitted to the packet-switching communications network instead of the first signaling data when the second signaling data cannot be converted to the first signaling data.

However, Ress teaches a method of protocol interworking where message tunneling is used to transfer a native protocol message (second signaling data) from one protocol agent to another protocol agent without converting to and from the agent interworking protocol (first signaling) in the case that the native protocol message does not map to the other agent protocol as spoken of on column 9, lines 6-16.

At the time of the invention, it would have been obvious to someone skilled in the art to combine the tunneling teachings of Ress with the interworking teachings of Rose in order to communicate messages or parameters which do not map to any other agent protocols, but provide added value for a call between two devices as spoken of on column 9, lines 6-16 of Ress.

Regarding claim **27**, Rose further teaches gateway interface 112 shown in Figure 5 contained within exchange 142.

Response to Arguments

7. Applicant's arguments with respect to amended claims **1, 21, 24, and 26** have been considered but are moot in view of the new ground(s) of rejection provided above.

Conclusion

8. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Berg et al. (U.S. 6,680,952) is another reference pertinent to this application.

9. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL.**

See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael J. Moore, Jr. whose telephone number is (571) 272-3168. The examiner can normally be reached on Monday-Friday (8:30am - 5:00pm).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Seema S. Rao can be reached at (571) 272-3174. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Michael J. Moore, Jr.
Examiner
Art Unit 2666

mjm MM

DANG TON
PRIMARY EXAMINER

Notice of References Cited	Application/Control No. 09/827,487	Applicant(s)/Patent Under Reexamination BRUMM ET AL.	
	Examiner Michael J. Moore, Jr.	Art Unit 2666	Page 1 of 1

U.S. PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Name	Classification
	A	US-6,885,658 B1	04-2005	Ress et al.	370/352
	B	US-6,680,952 B1	01-2004	Berg et al.	370/467
	C	US-			
	D	US-			
	E	US-			
	F	US-			
	G	US-			
	H	US-			
	I	US-			
	J	US-			
	K	US-			
	L	US-			
	M	US-			

FOREIGN PATENT DOCUMENTS

*		Document Number Country Code-Number-Kind Code	Date MM-YYYY	Country	Name	Classification
	N					
	O					
	P					
	Q					
	R					
	S					
	T					

NON-PATENT DOCUMENTS

*		Include as applicable: Author, Title Date, Publisher, Edition or Volume, Pertinent Pages)
	U	
	V	
	W	
	X	

*A copy of this reference is not being furnished with this Office action. (See MPEP § 707.05(a).)
Dates in MM-YYYY format are publication dates. Classifications may be US or foreign.

APPENDIX C

U.S. Patent No. 6,396,840 ("Rose et al.")

US006396840B1

(12) United States Patent
Rose et al.

(10) Patent No.: US 6,396,840 B1
(45) Date of Patent: *May 28, 2002

(54) METHOD, INTERFACE AND SYSTEM FOR CONNECTING COMMUNICATION TRAFFIC ACROSS AN INTERMEDIATE NETWORK

(75) Inventors: **Desne Jean Rose**, St Albans; **Roy Harold Mauger**, Radlett, both of (GB)

(73) Assignee: **Nortel Networks Limited**, St. Laurent (CA)

(*) Notice: This patent issued on a continued prosecution application filed under 37 CFR 1.53(d), and is subject to the twenty year patent term provisions of 35 U.S.C. 154(a)(2).

Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/089,796**

(22) Filed: **Jun. 3, 1998**

(30) **Foreign Application Priority Data**

Jun. 6, 1997 (GB) 9711788

(51) **Int. Cl.**[7] **H04J 3/02**

(52) **U.S. Cl.** **370/401**

(58) **Field of Search** 370/351, 352, 370/401, 400, 389, 399, 397, 396, 395, 465, 466, 468, 335, 537, 503, 229, 412, 516, 460, 252–255, 353, 360, 364, 394, 406, 409, 467, 469, 471, 474–476; 379/14, 16, 95.15, 93.14, 93.07, 93.05, 93.31, 219, 220, 225, 232, 240, 242, 229, 230, 231

(56) **References Cited**

U.S. PATENT DOCUMENTS

5,592,477 A * 1/1997 Farris et al. 370/396
5,914,934 A * 6/1999 Rathnavelu 370/229
5,923,659 A * 7/1999 Curry et al. 370/401

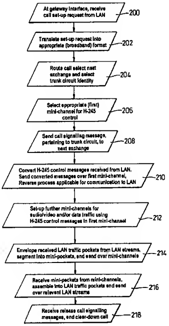
* cited by examiner

Primary Examiner—Dang Ton
(74) *Attorney, Agent, or Firm*—Lee, Mann, Smith, McWilliams Sweeney & Ohlson

(57) **ABSTRACT**

Interconnection of a multimedia terminal (**110**) of a narrowband, LAN-type network (**10**) to an exchange (**118**) and thence to an end-point (**119**) is orchestrated through an intermediate network (**142**), as shown in FIG. **5**. A route (**115**) to the exchange (**118**) is initially established by a call handler (**116**) in responsive to a called party number of the end-point, before a connection supervisor (**120**), coupled to the call handler (**116**), sets up a control channel across the intermediate network (**142**). The control channel supports the communication of control messages between the multimedia terminal (**110**) and the end-point (**119**), which control messages are intercepted and interpreted by the connection supervisor (**120**). The connection supervisor (**120**) then establishes media paths through the intermediate network (**142**) dependent upon types of control message sent across the control channel, which media paths are used to transfer traffic components across the intermediate network.

24 Claims, 6 Drawing Sheets



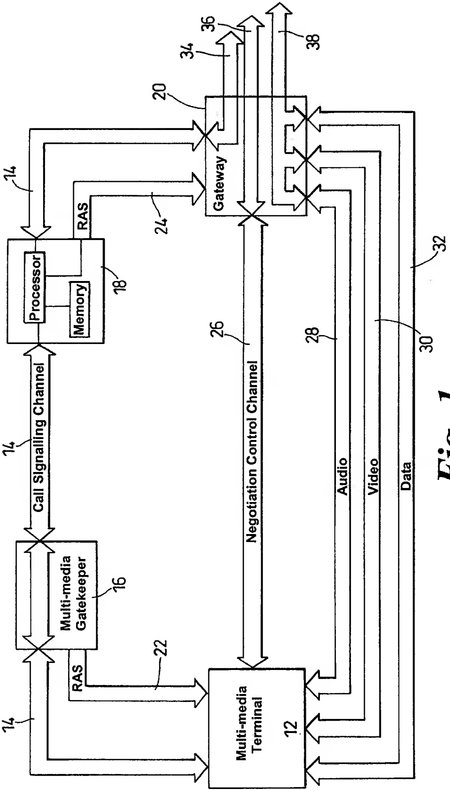


Fig. 1
Prior Art

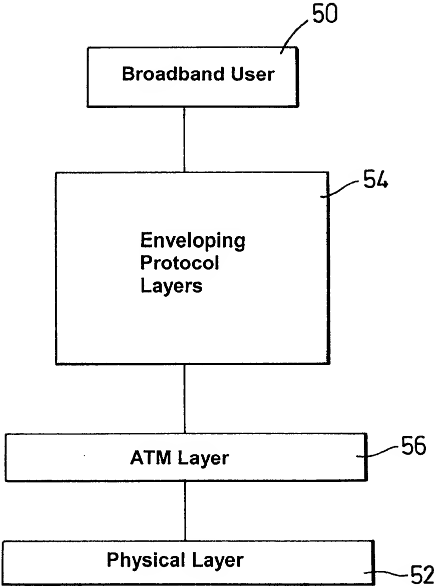


Fig. 2
Prior Art

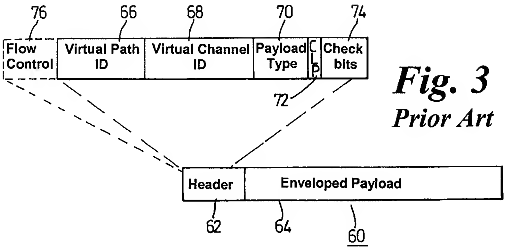


Fig. 3
Prior Art

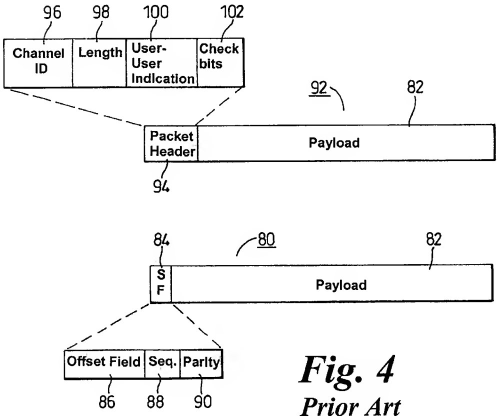


Fig. 4
Prior Art

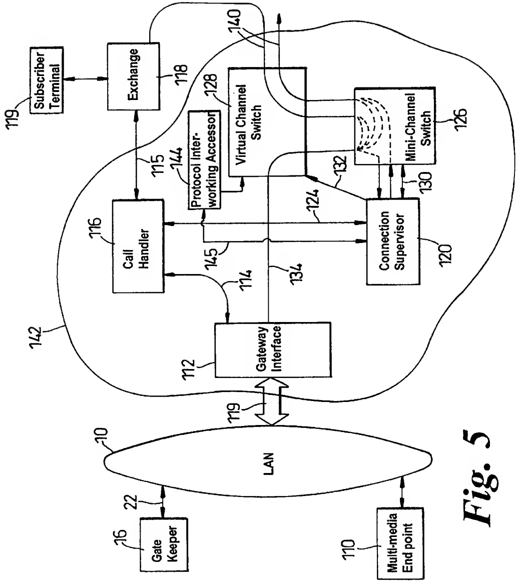

Subscriber Terminal
Exchange
Virtual Channel Switch
Protocol Inter-working Accessor
Mini-Channel Switch
Call Handler
Connection Supervisor
Gateway Interface
LAN
Gate Keeper
Multi-media End point
119
118
140
128
126
115
144
132
130
124
116
145
114
134
120
142
112
10
22
16
110

Fig. 5

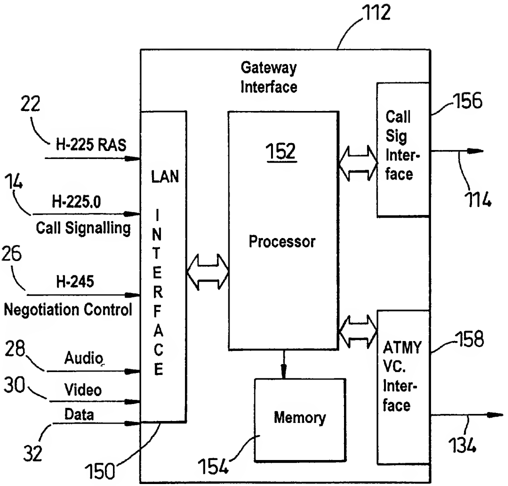

112
Gateway
Interface
22
H-225 RAS
14
H-225.0
Call Signalling
26
H-245
Negotiation Control
28
Audio
30
Video
Data
32
150
LAN
INTERFACE
152
Processor
Memory
154
Call
Sig
Inter-
face
156
114
ATMY
VC.
Inter-
face
158
134

Fig. 6

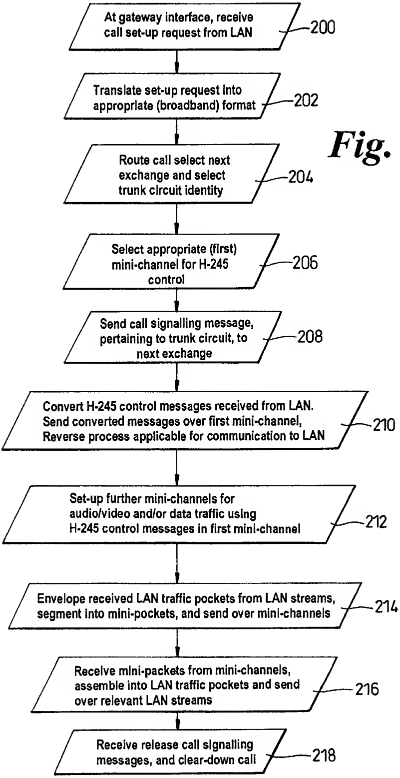

At gateway interface, receive call set-up request from LAN
200
Translate set-up request into appropriate (broadband) format
202
Route call select next exchange and select trunk circuit identity
204
Select appropriate (first) mini-channel for H-245 control
206
Send call signalling message, pertaining to trunk circuit, to next exchange
208
Convert H-245 control messages received from LAN. Send converted messages over first mini-channel, Reverse process applicable for communication to LAN
210
Set-up further mini-channels for audio/video and/or data traffic using H-245 control messages in first mini-channel
212
Envelope received LAN traffic pockets from LAN streams, segment into mini-pockets, and send over mini-channels
214
Receive mini-packets from mini-channels, assemble into LAN traffic pockets and send over relevant LAN streams
216
Receive release call signalling messages, and clear-down call
218

Fig. 7

METHOD, INTERFACE AND SYSTEM FOR CONNECTING COMMUNICATION TRAFFIC ACROSS AN INTERMEDIATE NETWORK

This application claims priority from United Kingdom Application No.: 97117881 filed Jun. 6, 1997 in the name of Northern Telecom Limited.

BACKGROUND OF THE INVENTION

This invention relates, generally, to a communication system architecture and operating protocol therefor, and is particularly, but not exclusively, applicable to an interface arrangement that integrates a local area network (LAN), typically operating in a wide-band context, with a broadband virtual circuit-switched system, such as envisaged and implemented in Asynchronous Transmission Mode (ATM) networks.

SUMMARY OF THE PRIOR ART

Telephony systems have evolved from simplistic hard-wired interconnected networks to broadband, high capacity systems that support multimedia, multi-mode communication devices on local area networks (LANS) and packet-switched communication systems. Indeed, instead of having to rely entirely on dedicated land-line infrastructure, present day technologies now occupy virtual channel environments in both the radio frequency and land-line domains.

The designers of today's narrowband communication systems, which typically employ pulse code modulation at a data rate of 64 kilo-bits per second (kbps), are presently considering the adaptation and development of these narrowband communication systems to support a migration to a multimedia environment having data rates of two (2) Mega-bits per second (Mbps) and beyond. As will be understood, the requirement for migration arises as a direct consequence of the vast costs involved in deploying global communication systems, with ATM being touted as providing a low cost and simple package that is capable of supporting migration from narrowband (or wide-band) to broadband applications (principally in the intervening period before the full deployment of a free-standing Universal Mobile Telecommunication System (UMTS), for example).

It has also been necessary for designers to consider and anticipate the extensive and elaborate requirements for future control signalling and call management techniques. In this respect, new signalling schemes, such as AAL-2 negotiation procedures, have been developed to provide robust, high bandwidth communications at high data rates, while designers have also been keen to define system architectures in terms of "stacks" that comprise discrete layers of infrastructure or signalling protocols that each add functionality, capacity or control over a preceding layer in the stack.

The problems faced by system designers are further exacerbated by the fact that, to date, the various different forms of communication system, e.g. ATM, LANs and cellular radiotelephone schemes, operate distinct signalling and transport protocols that are incompatible on a network-to-network basis.

GB-A-2311690 describes the merging of two networks in which a telephone subsystem is connected to a packet-switched broadband backbone and in which telephony is added to the backbone without interfering with packetised data. GB-A-2309362 is a mechanism for interconnecting broadband and narrowband networks and is generally related to the present field of the present invention.

WO 96/06492 is an arrangement for supplying local network emulation service over a public connectionless ATM network. More specifically, a server acts as an address resolver and as a relay for routing traffic. SynOptics U.S. Pat. No. 5,420,858 describes the segmentation and re-assembly of information between non-ATM messages and ATM cells.

U.S. Pat. No. 5528590 describes the transfer of data between an ATM-UNI interface and an ATM-LAN interface in a manner such that the ATM-UNI interface recognises frames and assembles and ATM cells into these frames. More particularly, the system can determine whether or not there is enough capacity on the LAN interface for the frame, and only if there is enough capacity is the frame transferred via a ATM switch to the ATM-LAN interface and then onwards to the LAN.

It is therefore clearly desirable to design and produce a communication system architecture that supports varying types of present-day communication network, with the communication system architecture at least possessing an interface that has the capability of handling broadband signalling and transport schemes and which also contemplates the interconnection of LAN or WAN architectures to such broadband networks.

SUMMARY OF THE INVENTION

In a first aspect of the present invention there is provided a method of connecting a first network to a second network via an intermediate network, the first network and second network using a set of control messages to control media paths between the first network and the second network, the method comprising the steps of: establishing a control channel across the intermediate network to carry the set of control messages; intercepting the set of control messages in the intermediate network and determining a requirement for media paths in response thereto; in response to the determination, setting up media paths in the intermediate network to connect paths to carry media traffic between the first network and the second network.

In another aspect of the present invention there is provided a method of connecting communication traffic comprised of a plurality traffic components across a broadband network from a local area network, the method comprising the steps of: in the local area network, generating control messages for controlling the traffic components and applying these control messages to an interface of the broadband network; establishing a communication path within the broadband network to carry at least one of the plurality of traffic components; and in the broadband network, using the control messages to control transfer of the plurality of traffic components over the communication path.

In another aspect of the present invention there is provided a method of interconnecting communication traffic across a broadband network from a local area network (LAN), the broadband network having a transfer protocol that supports mini-channels in a virtual circuit-switched environment, the LAN (**10**) providing the communication traffic as LAN streams to an interface of the broadband network, the method comprising the step of mapping the LAN streams to the mini-channels.

In a preferred embodiment, the LAN streams include audio, video, data and control streams, and the method further comprising the step of interpreting the control streams to set-up mini-channels used to carry at least one of an audio, video and a data communication.

In yet another aspect of the present invention there is provided a connection supervisor for orchestrating the com-

munication of traffic components between first and second networks via an intermediate network, the connection supervisor responsive to control messages communicated between the first and second networks, the connection supervisor including: means for setting-up a communication path for carrying the control messages across the intermediate network; means for determining types of control message sent across the communication path; and means for establishing media paths dependent upon types of control message sent across the communication path, the media paths arranged to transfer the traffic components across the intermediate network.

In still yet another aspect of the present invention there is provided a communication node having a gateway that provides an interfaces to a first end-point in a network, the first end-point arranged to initiate a call through the communication node by sending to the gateway a called party number of a second end-point coupled to an exchange and wherein control messages are communicated between the first end-point and the second end-point, the communication node further comprising: a call handler coupled to the gateway and responsive to the called party number, the call handler arranged to select a route to the exchange; and a connection supervisor, coupled to the call handler and operationally responsive thereto, the connection supervisor having: i) means to set-up a control channel that supports transfer of the control messages between the gateway and the exchange in response to the call handler receiving the called party number; ii) means for determining types of control message sent across the control channel; and iii) means for establishing media paths between the gateway and the exchange (118) dependent upon types of control message sent across the control channel, the media paths arranged to transfer traffic components across the communication node.

In a preferred embodiment, the communication node is a broadband network and the control channel and the media paths are virtual channels.

Beneficially, the preferred embodiments of the present invention generally provide an ability of interconnecting a first LAN-compatible system (such as a WAN) through a seamless public or private broadband network (supporting narrowband or broadband telephony) to another LAN-compatible system.

BRIEF DESCRIPTION OF THE DRAWINGS

Exemplary embodiments and aspects of the present invention will now be described with reference to the accompanying drawings, in which:

FIG. 1 is a block diagram of a prior art local area network, such as implemented in an H.323 Ethernet architecture;

FIG. 2 illustrates the concept of an architectural stack, typically employed within a prior art broadband network;

FIG. 3 illustrates a data frame structure for a prior art ATM network;

FIG. 4 illustrates a typical frame arrangement used for enveloping data into the data frame structure of FIG. 3;

FIG. 5 is a block diagram of an integrated system architecture, according to a preferred embodiment of the present invention, for an interconnected broadband-LAN environment;

FIG. 6 represents a block diagram of a gateway of FIG. 5, the gateway constructed according to the preferred embodiment of the present invention; and

FIG. 7 is a flow diagram illustrating how, in accordance with a preferred method of the present invention, call set-up is established within the system of FIG. 5.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to FIG. 1, there is shown a block diagram of a prior art local area network (LAN) **10** suitable for supporting an Ethernet connection regime, or the like. The LAN **10**, as will be appreciated, operates in a bursty fashion and provides packets of data over an H.323 signalling scheme, or similar messaging protocol. As will be understood, the H.323 signalling scheme defines the functionality of the multimedia terminal **12**, the signalling protocols utilised within the LAN **10**, the types of terminals suitable for use with the LAN **10** and the transmission protocols adopted for use by the multimedia terminal **12**. Although, for the sake of clarity, only a solitary multimedia terminal **12** is shown connected within the LAN **10**, it will be appreciated that the LAN **10** can support a multitude of multimedia terminals offering differing levels of functionality to each user thereof.

As will be appreciated, in a LAN environment a limited bandwidth supports numerous packet-based communications that vie for the available bandwidth. When using H.323 protocols over the LAN architecture, port addresses of a first end point are associated with port addresses of a second end point, with the resultant interconnection between pairs of port addresses referred to (generally) as an H.323 channel. In this context, the term "end point" relates to a terminal, a gatekeeper or a gateway (the functions of which will be described later). Each H.323 video or audio channel can be a wideband channel presently supporting data up to a rate of 2 Mbps.

As will be understood, the multimedia terminal **12** and the multimedia gateway **20** each have unique port addresses through which communication (interconnection) is established. Each port address is typically comprised of the LAN address and a port number, with the LAN address usually common to a specific piece of equipment (i.e. the gateway **20** or a multimedia terminal).

A dedicated call signalling channel **14** couples the multimedia terminal **12** to a first multimedia gatekeeper **16**, which first multimedia gatekeeper **16** is, in turn, coupled to a second multimedia gatekeeper **18** through the call signalling channel **14**. The second multimedia gatekeeper **18** is further coupled to a multimedia gateway **20** (or "multimedia termination point", such as a printer) through the call signalling channel **14**. Both the first multimedia gatekeeper **16** and the second multimedia gatekeeper **18** are, respectively, coupled to the multimedia terminal **12** and the multimedia gateway **20** via a registration, admission and status (RAS) channel **22**–**24**. The call signalling channel uses the H.323 signalling protocol. In the context of the prior art, the use of either or both gatekeepers is optional and is included for a more complete understanding of a set-up of a H.323 call.

The function of the multimedia gatekeeper, as will be appreciated, is principally to translate LAN addresses into appropriate network addresses, and to negotiate and control bandwidth requirements for a proposed H.323 communication. Specifically, in response to the multimedia terminal **12** generating an alias network address (i.e. not a LAN address, but something like an e-mail address), the gatekeeper operates to translate the alias address into a usable network or LAN address. More particularly, a processor in the gatekeeper will typically access a look-up table (shown only in relation to the second gatekeeper **18** for clarity) to ascertain the usable network or LAN address, whereafter the gatekeeper updates the multimedia terminal **12** with the usable network or LAN address via the RAS **22**. The network

address is analogous to a telephone number in a conventional telephone system, although the network address may be formulated in such a way that it can address multiple terminals simultaneously.

It will be understood that the multimedia gatekeepers **16–18** may be co-located with the multimedia terminal **12** and the multimedia gateway **20**, and are illustrated as distinct blocks for the sake of explanation. While the LAN is described as having a multimedia gateway **20** (that provides access to different networks having different signalling protocols via a signalling channel resource **34**, a control channel resource **36** and channels **38** that support audio, video and/or data), the gateway **20** could be substituted for a second multimedia terminal or a multi-point control unit (namely a conference bridge).

The LAN **10** operates with three principal signalling schemes for each multimedia call. The purpose and function of these schemes will now be described.

Call signalling information is communicated along the call signalling channel **14** and is arranged, principally, to set-up and clear-down calls. Call signalling information generally includes routing information (e.g. the network or LAN address), acknowledge back signalling, connection request/release instructions and input/output port addresses. Assuming that a suitable network address is eventually output from an end point, e.g. multimedia terminal **12**, the network address is passed along the call signalling channel **14** and routed via at least the first multimedia gatekeeper **16** (and probably he second multimedia gatekeeper **18**) to a receiving end point, e.g. the multimedia gateway **20**. More particularly, the network address is typically encoded in a set-up message, as will readily be appreciated, and also identifies the port for the negotiation control channel **26** that the multimedia terminal **12** intends to use. The set-up message, sent from the multimedia terminal **12**, causes the receiving unit (in this example, the gateway **20**) to respond by sending a port identification and LAN terminal address over the call signalling channel **14**. In this way, the receiving unit (in this case the multimedia gateway **20**) identifies to the multimedia terminal **12** which port the receiving unit intends to use for the negotiation control channel **26**. As such, both the requesting multimedia terminal **12** and the called party each possess an address of a port to which communications on the LAN **10** are to be directed.

Once an understanding (in terms of port usage) has been established between parties that are to participate in the communication, the call signalling channel **14** is used to administer overall system control, while the negotiation control channel **26** (established between the identified port addresses) is used for two principal purposes. First, the negotiation control channel **26** is used to communicate in-call channel information, such as timing information, channel frequency information, data rates and bandwidth allocations. Secondly, the negotiation control channel **26** is used to identify the port addresses (at all terminals) and to control transmissions on the audio stream **28**, video stream **30** and data stream **32**. The negotiation control channel **26** may utilise H.245 signalling or the like.

In an alternative prior art system, namely a broadband network, it will be appreciated that, conceptually, the systems architecture can be considered to comprise discrete architectural layers; this is illustrated in detail in FIG. **2**. Specifically, broadband networks, such as those which utilise ATM, are derived from circuit switched telephony and so typically exhibit several intermediate signalling layers between a broadband user **50** and a physical infrastructure layer **52**. More particularly, there is usually at least one intermediate enveloping protocol layer **54** juxtaposed to the broadband user **50**, while an ATM (packet-switched) signalling protocol layer **56** is sandwiched between the physical infrastructure layer **52** and the enveloping protocol layer **54**. Consequently, user information provided by the broadband user **50** is first packaged into defined protocol envelopes (by the enveloping protocol layer **54**), which envelopes are then compressed into a packet-switched format by the ATM signalling protocol layer **56**. Once fully packaged, information can be transmitted across the broadband network through the physical layer **52**.

Therefore, unlike narrowband networks, i.e. circuit-switched communications having a fixed amount of bandwidth per channel, that provide a continuous transmission of information (even in the context of time division multiplexed communication), a broadband network utilises a transfer protocol in which virtual channels are circuit-switched and which provides a provisioned (but varying) bandwidth. Broadband networks can utilise ATM and AAL-2 (ATM Adaptation Layer 2); the latter is a subset of ATM that provides switching at a virtual sub-channel level in an ATM environment. Other protocols used within ATM include AAL-1 and AAL-5. AAL-1 is an ATM adaptation protocol originally targeted at constant bit rate (CBR) traffic, e.g. voice or video, and is applicable to data rates equal to or exceeding sixty-four kbps. AAL-5 provides a capability of data, voice and video transmissions to work stations, and is therefore particularly applicable to multimedia communication systems. AAL-5 segments long data structures into many cells, with a data structure conceivably exceeding fifteen hundred octets in length.

Turning now to FIG. **3**, there is shown basic cell frame structure **60** of a prior art broadband network. For the purpose of explanation, if we now consider the data frame structure **60** as being suitable for ATM transmission, the data frame structure **60** comprises a header **62** of control information and an enveloped payload **64**. The header **62** comprises a virtual path identifier **66** and a virtual channel identifier **68** that together co-operate to identify a circuit-switched path (i.e. a virtual channel) through the broadband network. The circuit-switched path is therefore set at the beginning of a call and only released at the end of the call. The header **62** further includes an indication of payload type **70**, and an indication termed cell loss priority **72** that stipulates whether the communication on the virtual channel can be dropped to support higher priority communications. As will be appreciated, there is a finite amount of capacity offered by the broadband network and so it may occasionally be necessary to consider the voluntary release of channel resources. Finally, the header **62** includes check-bits for error detection and correction, although the header **62** may optionally include dedicated flow control bits **76** used in quasi-broadband systems to enhance data rate capacity over existing communication resources, e.g. by superimposing high frequency channels over an existing two-wire scheme. More particularly, the generic flow control bits act as negotiation bits and request the assignment of bandwidth, for example, from a system controller (not shown).

Use of this form of packet-switched structure therefore allows interleaving of packets across a shared physical resource, albeit that a virtual channel used for the communication is unique to that communication. The enveloped payload **64**, which is of fixed length, will now be described in more detail in relation to FIG. **4** in which there is shown a typical mechanism by which data is "nested" within the payload envelope **64** of FIG. **3**. Particularly, data that is

ultimately to be nested within the payload envelope **64** can vary in length, and can be comprised from distinct data portions. Indeed, a combination of the individual data portions can produce a data string having an overall length that exceeds the length of the payload envelope **64**. Consequently, the data may be encoded using known techniques so as to optimise nesting of the data into the payload envelope **64**.

In relation to an AAL-2 protocol data unit (PDU) **80**, data **82** is preceded by a start-field octet **84** comprising an offset field **86**, a sequence number **88** and parity bit **90**. Alternatively, with respect to an AAL-2 service data unit (SDU) **92**, the data **82** (which, in this instance, usually varies in length) is preceded by a packet header **94** comprising a channel identifier **96**, a length indicator **98**, a user-to-user indication **100** and check bits **102**. The channel identifier **96** identifies a "mini-channel" that uniquely supports a solitary communication. As such, more than one mini-channel can be nested or interleaved within a single enveloped ATM cell payload **64** of FIG. **3**. The length indicator **98** identifies the length of the data portion. The functions of the constituent parts of the packet header **94** are detailed in ITU standards document 1.363.2

As will now be appreciated, the exemplary combination of FIG. **3** and FIG. **4** demonstrate the stack concept illustrated in FIG. **2**. The PDU and SDU layers for AAL-1 and AAL-5 vary from the structure of AAL-2, but both form stacks within ATM in a similar fashion to that described above, as will be readily appreciated.

Referring now to FIG. **5**, a preferred embodiment of the present invention is shown. The present invention provides a mechanism for the interconnection of a LAN to a broadband network, perhaps implemented using ATM. In relation to the figure, elements common with the prior art contain identical reference numerals to those of the earlier drawing figures.

The LAN **10**, as previously described, provides a capability of interconnecting communication devices (i.e. multimedia endpoints **110**), such as computers (having Internet capabilities) and multimedia terminals **12** and other multimedia devices. As in a conventional system, the LAN **10** may also support a gatekeeper **16**. It will be appreciated that a communication resource **111**, coupled to a gateway interface circuit **112**, supports the transmission of RAS bits and provides a dedicated call signalling channel, a dedicated negotiation control channel and audio, video and data streams (as previously described and shown in relation to FIG. **2**, albeit not specifically shown in this drawing figure).

The gateway interface circuit **112** couples call signalling messages **114** to a call handler **116**, typically arranged to support an integrated service digital network (ISDN) methodology (either narrowband, broadband or a hybrid). The call signalling messages **114** are used to set-up and clear-down calls, and are also used to identify multimedia terminal addresses and the like. The call handler **116** is, in turn, coupled to a succession of other exchanges **118** through a semi-permanent call signalling channel **115**. At least one subscriber terminal **119** is coupled to each other exchange, with the subscriber terminal **119** having a unique address. The connection supervisor **120** is connected through a control line **124** to the call handler **116**.

The connection supervisor **120** is arranged to supervise the control of both a mini-channel switch **126** and a virtual channel switch **128** via control lines **130** and **132**, respectively. The virtual channel switch **128** is coupled to the gateway interface **112** via a first virtual channel resource **134** supporting (in the exemplary context of AAL-2) enveloped mini-channel payloads, e.g. H.245 negotiation control messages, and audio, video or data packets. Before providing an output on a second channel resource **140**, the virtual channel switch **128** routes the payloads received on the first virtual channel resource **134** through the mini-channel switch **126**, which mini-channel switch **126** is arranged to optimise call transmissions ultimately output by the virtual channel switch **128** on the second virtual channel resource **140**. The second virtual channel resource **140** leads to the other exchange **118**.

The connection supervisor **120** provides a dual function. First, it acts to control the virtual channel switch **128** (via control line **132**), and the mini-channel switch **126** (via control line **130**). Second, the connection supervisor **120** also functions to receive, process and generate H.245 messages for H.323 calls. In this latter respect, H.245 messages are routed between the first virtual channel resource **134** and the connection supervisor **120** and also between the connection supervisor **120** and the second virtual channel resource **140**, with both routings being via the virtual channel switch **128** and the mini-channel switch **126**.

The gateway interface **112**, the call handler **116**, the connection supervisor **120**, the virtual channel switch **128** and the mini-channel switch **126** constitute parts of an exchange (or node) **142**.

The present invention also has application in relation to AAL-1 and AAL-5, which operational embodiments will be described in more detail later. However, to support hybrid working between AAL-1, AAL-2 and AAL-5 the exchange **142** further includes a protocol interworking processor **144** that translates between AAL-1, AAL-2 and AAL-5. This protocol interworking processor **144** is coupled to the virtual channel switch **128**. The protocol interworking processor **144** is operationally responsive to the connection supervisor **120** (via control line **145**). One will appreciate that the mini-channel switch **126** is not required in relation to AAL-1 and AAL-5 specific calls. H.245 messages carried on AAL-5 instead of AAL-2 are routed solely through the virtual channel switch and through the connection supervisor; this connection is not shown for the sake of clarity of FIG. **5**.

FIG. **6** illustrates the structure of the gateway interface **112** in greater detail and also according to a preferred embodiment of the present invention. The gateway interface **112** is responsive to a LAN **10** and receives, at LAN interface **150**, an H.225.0 RAS control channel **22**, an H.225.0 call signalling channel **14**, an H.245 negotiation control channel **26** and audio streams **28**, video streams **30** and data streams **32**. A processor **152**, coupled to a memory device **154**, controls the routing of the various input channels and streams (applied to the LAN interface **150**) to appropriate output interfaces.

A call signalling interface **156** receives a translated version of signalling messages received on the H.225 call signalling channel **14**, i.e. the processor **152** and memory device **154** co-operate to translate incoming call signalling messages into an acceptable broadband format, such as DSS1/IDSS2, for onward routing (via the control signalling channel **114**) to the call handler **116**. The processor **152** also packages control messages (received on the negotiation control channel **26**) and information (received on the audio, video and data streams **28–32**) into a mini-channel format suitable for use in the broadband network. This mini-channel format is output through a broadband ATM/virtual channel interface **158** to the first virtual channel resource **134**.

As will now be appreciated, the memory device **154** acts as a storage medium for temporarily storing information

passing between the LAN and a broadband network, and also contains look-up tables associated with address and routing information, active call and connection information, and signalling protocol translation schemes used to translate LAN signalling to narrowband/broadband signalling.

Operation of the architecture of the preferred embodiment of the present invention will now be described with particular regard to FIG. 7. In response to receiving conventional LAN streams from the call signalling channel **14** (step **200** of FIG. 7), the gateway interface **112** first converts call signalling information (received on the call signalling channel **14**) into an appropriate format, such as DSS1, and forwards this onward to the call handler **116**. More particularly, as will now be understood, the call signalling information contains an address of a called party (normally as a telephone number, although an E-mail address can also be used) and an identity (e.g. a telephone number and/or E-mail address) of a requesting multimedia terminal. As such, it might be necessary to translate (at least) the address of the called party into a format acceptable to the broadband network (step **202**). In other words, the gateway interface may need to generate a telephone number for use in the broadband network.

This address mapping process can be executed within the call hander **116** or within the gateway interface **112**, after which the communication system begins to establish a connection. As a consequence of this procedure, data received by the gateway interface **112** (by way of the audio, video and data streams **28–32** and the negotiation control channel **26**) will typically need to be stored, temporarily, in memory **154**. As will be appreciated, in a multimedia call, the LAN streams can be considered as forming distinct traffic components in the call.

Using the telephone number of the called party, the call handler selects an outgoing route, i.e. the next exchange **118**, and a trunk circuit leading to that next exchange (step **204**). The connection supervisor **120** is then notified of the selected trunk circuit. Optionally, the call handler can send an SS7 IAM to the next exchange **118** (via the call signalling channel **115**), but there is an associated risk because, at this time, there is no guarantee that a successful path can be set up across exchange **142**. In the event that an IAM is sent, then the relevant next exchange **118** then responds to the call handler **116** and identifies/confirms the address identity or identities that, respectively, has or have been ear-marked for the call; this mechanism is therefore analogous to the prior art procedure described in relation to FIG. **1**. The call handler **116** sends the identity of a selected trunk circuit to the connection supervisor **120** which in turn makes the connections across the virtual channel switch **128** and mini-channel switch **126** (as appropriate) to connect the H.245 control channel on the first virtual channel resource **134** to the connection supervisor **120** and then onto the second virtual channel resource **140** (step **206**). In this respect, the call handler is under the impression that it is setting up a whole trunk call whereas, in fact, the call handler **116** is only setting up the H.245 negotiation control channel.

As a brief re-cap, the calling party dials the number of the called party and, in response thereto, the call handler **116** analyses the called number and selects out-going route (based on the called number) to next exchange **118**. Preferably, the call handler **116** selects a trunk circuit belonging to the out-going route, although this function may be performed by the connection supervisor **120**. Rather than asking the virtual channel switch **128** to set-up media paths for the call, the call handler **116** then asks connection supervisor **120** to set-up the call.

Step **206** is now described in more detail. The connection supervisor **120** interacts with the gateway interface **112**, the virtual channel switch **128** and the mini-channel switch **126** to orchestrate a broadband connection. A first step requires the selection of a first mini-channel of the first virtual channel resource **134**, which mini-channel is incident to the gateway interface **112**. Preferably, the connection supervisor **120** makes the selection of the first mini-channel. A first connection is made (through use of control channels **130–132**) between the gateway interface **112** and the connection supervisor **120**, which connection uses the first mini-channel and is made via the virtual channel switch **128** and the mini-channel switch **126**. The connection supervisor then uses the trunk circuit identity (received from the call handler **116**) to select a virtual channel and a second mini-channel from the available virtual channels of the second virtual channel resource **140**. A second connection is then made between the connection supervisor **120** and the other exchange **118** using the selected virtual channel and the second mini-channel via the virtual channel switch **128** and the mini-channel switch **126**. The connection supervisor **120** associates the first mini-channel and the second mini-channel with each other and the H.323 call.

At step **208**, the call handler **116** sends a signalling message over the call signalling channel **115** to provide details of the set-up to the next exchange **118**. In the preferred embodiment, the signalling message is an SS7 IAM containing the selected trunk circuit identity, the virtual channel identity and the mini-channel identity; the latter two are within the user-to-user field. The call handler **116** should receive from the next exchange **118** a message confirming the trunk circuit identity, etc. However, if an IAM was sent during step **204** (and hence did not include the virtual channel identity and mini-channel identity), then the virtual channel identity and the mini-channel identity must now be sent within a SS7 user-to-user information message.

The initial communication with the next exchange can actually be performed within step **204** or within step **208**; the latter is a safer mechanism because the path has been established to the next exchange at this point.

The connection supervisor **120** instructs the gateway interface **112** to launch any previously stored H.245 control messages (received on the negotiation control channel **26**) to the first mini-channel that has just been set up. Specifically, the stored control messages are formatted into packets and cells as required by the mini-channels, and then placed on the ATM virtual channel **134** for transmission to the connection supervisor (step **210** of FIG. **7**) and then onto the next exchange **118** via the second mini-channel. Furthermore, using H.245, the end points (in this case multimedia terminal **110** and subscriber terminal **119**) exchange control messages via the connection supervisor **120** to ascertain a common functional capability regarding audio, video and data.

The call handler **116** is now under the impression that the call set-up has been completed.

The next stage is to set up the required audio, video and/or data paths. Typically (but not necessarily), all mini-channels for the same H.323 call reside within a single virtual channel. In relation to each required path, the following applies.

In step **212**, the calling unit that initiated the call set-up (i.e. the multimedia end point **110** in this example) now sends an H.245 control message to the exchange **142**, which message is actually relayed to the connection supervisor **120**. The connection supervisor **120** assimilates the infor-

mation contained in the H.245 control message and sets up a path between the gateway interface **112** and the next exchange **118**. To accomplish such a path, the connection supervisor **120** selects: i) a third mini-channel of the first virtual channel resource **134**; and ii) a fourth mini-channel of the second virtual channel resource **140**. The connection supervisor **120** then connects the third min-channel and the fourth mini-channel together via the virtual channel switch **128** and the mini-channel switch **126**. The connection supervisor **120** generates relevant H.245 control messages and sends them to the next exchange **118**. Upon receipt of H.245 control messages from the next exchange **118**, the connection supervisor **120** sends the corresponding H.245 control messages back to the gateway interfacc **112** for transmission back to the multimedia end-point **110**.

The process described above must be repeated for every audio, video or data path required.

The gateway interface **112** now operates to encode any stored traffic (obtained from the audio, video and data streams) into mini-channels that are then communicated to the next exchange **118** and ultimately (in an appropriate form) to the subscriber terminal **119**. As will be understood, the initiating end-point may start to transmit information before the exchange **142** (as a whole) is quite ready. Therefore, buffering is usually provided within the gateway interface **112**.

At step **214** of FIG. **7**, audio, video and/or data transmission can now occur over the assigned mini-channels set up for these purposes. In relation to the LAN streams, LAN traffic packets from the respective streams must be segmented (i.e. sized and labelled with a header) into mini-packets (e.g. AAL-2 packets). In the reverse direction, mini-packets are re-assembled to form LAN packets for the respective LAN streams (step **216**).

The set-up of the H.323 call is now complete.

There are numerous ways of clearing down the H.323 call. It is possible to have a partial clear-down in which audio, video or data paths are individually cleared down. To do this, an H245 control message is sent to the connection supervisor **120** that reacts by clearing down the relevant mini-channels. Alternatively, the whole call can be released by sending a release message over the call signalling channel **114** or **115** to the call handler **116**. The call handler is unable to clear down the call itself and must therefore solicit the assistance of the connection supervisor **120** to clear down all mini-channels related to the H.323 call. The mechanism is, however, dependent upon the direction from which clear down is initiated. Specifically, different signalling systems exist between: the call handler **116** and the gateway interface (e.g. DSS1 or DSS2); and the call handler **116** and the next exchange **118** (e.g. signalling system no. 7 (SS7)).

In relation to the operation of the mini-channel switch **126**, the connection supervisor **120** is responsible for associating the input and output ports of the mini-channel switch **126** and therefore accordingly notifies the mini-channel switch **126**.

To describe the invention is a different but complementary way, one can consider the following. Call signalling is used to set-up and clear-down an H.245 control channel applied to the gateway interface **112**. On the LAN **10**, call signalling is achieved using H.323 (H.225) call signalling messages; while DSS1/DSS2 signalling mcssages arc utiliscd in thc narrowband/broadband access network, and SS7 N-ISUP/B-ISUP signalling messages are used for call signalling in the narrowband/broadband trunk network. On the LAN **10**, routing of the H.323 call can be based upon transport addresses, telephone numbers (as per E-164) or E-mail addresses, while the call handler **116** bases its routing upon telephone numbers. Also, on the LAN **10** and where appropriate, the relevant infrastructure and subscriber entities know the transport address of each end of the H.245 control channel, whereas a relevant call handler in thc acccss network knows the access circuit identity for the H.323 call. In the trunk network, the relevant call handler knows the trunk circuit identity used for the H.323 call.

In other words, the call handler **116** has been hood-winked in the present invention into believing that the gateway interface **112** is a subscriber and hence operating within its access network. The call handler **116** believes that the next exchange **118** is connected to its trunk network (either narrowband or broadband).

When the call handler **116** sets up an H.323 call, the call handler **116** believes that the whole call has been established while, in fact, only the H.245 control channel has been set up. In the system of the present invention, no call handler or call signalling message knows the identity of any audio, video or data channel.

An outgoing call from the LAN **10** will be established on the following basis. The first significant event occurs when the call handler **116** receives a DSS1/DDS2 set-up message from the gateway interface **112**. In response thereto, the call handler **116** performs digit analysis (of the called telephone number) and then selects an outgoing route (and hence a next exchange) while also selecting a trunk circuit within the outgoing route. The outgoing route must be selected before any inter-exchange virtual channel can be selected. The connection supervisor **120** obtains the outgoing trunk circuit identity from the call handler **116** and then selects and sets up associated virtual channels and mini-channels on which control messages will be sent and received.

In relation to the bandwidth of an outgoing call, a bearer capability field in the H.323 call signalling set-up message, received from the LAN **10**, indicates the required bandwidth for the call. This bandwidth indication is then used by the connection supervisor **120** to select a virtual channel of appropriate bandwidth between the gateway interface **112** and the virtual channel switch **128**. Usually, subsequent virtual channels used for the H.323 call will have the same bandwidth.

For an incoming call, the call handler **116** receives, from an interconnected exchange **118**, an SS7 N-ISUP/B-ISUP IAM message on the call signalling channel **115**. This message contains a trunk circuit identity associated with an H.245 control mini-channel. The IAM message also includes, within its user-to-user field, an indication of which mini-channel in which incoming virtual channel (used to relay H.245 control messages) corresponds to the above mentioned trunk circuit identity; this indication is utilised by the connection supervisor **120** to identify the appropriate virtual channels and mini-channels. The call handler **116** asks the connection supervisor **120** to set up a single 64 kbps circuit (in the narrowband case), i.e. the circuit required for use as the H.245 control channel. Note that, in a preferred embodiment, the connection supervisor is arranged to set up an appropriate virtual channel and mini-channel leading to the gateway interface **112**, rather than a 64 kbps circuit. In relation to bandwidth allocation for an incoming call, the true requircd bandwidth will be obtaincd from thc user-to-user field of the IAM message. The connection supervisor then uses this bandwidth to set-up the appropriate virtual channel.

In relation to point-to-multi-point communication (which is supported by H.323), the connection supervisor **120** is arranged to consolidate separate calls (that would otherwise be supported on separate and distinct virtual channels) through a conference bridge connected to the mini-channel switch **126**.

In summary, therefore, once the relevant end-point (or terminal) identities (e.g. telephone numbers, E-mail addresses, etc.) and address identities (e.g. trunk circuit identity and virtual channel plus mini-channel identities) have been exchanged between the gateway interface **112** and the exchange **118**, a first AAL-2 mini-channel is used as a control (signalling) channel, and this first mini-channel then controls the setting up and clearing down of other AAL-2 mini-channels which support the same H.323 multimedia call between the multimedia endpoint **110** (of the LAN **10**) and the subscriber terminal **119**. In other words, H.323 LAN streams are converted into AAL-2 mini-channels by the gateway interface **112**, and then carried on a virtual channel which is itself controlled by an AAL-2 mini-channel using encoded H.245 control messages.

Basically, the present invention uses control messages specific to a first type of network in a different context within an intermediate network (i.e. a broadband network) such as to set-up requisite media paths in the intermediate network, whereas the prior art uses a gateway at each boundary to the intermediate network to convert entirely all control messages and media formats for transport across the intermediate network.

Rather than having the system of the present invention establish a trunk connection between the LAN and the called subscriber's exchange, the preferred embodiment of the present invention establishes AAL-2 mini-channels.

In relation to the application of the set-up procedure of the preferred embodiment, this set-up procedure is equally applicable, for example, to situations where AAL-5 is used instead of AAL-2, or to where a mixture of AAL-1, AAL-5 and AAL-2 are used instead of just AAL-2. It will be appreciated that the various ATM adaptation layers are geared towards optimal transport of different types of information and that, as such, AAL-2 is more efficient in relation to voice communication as compared with AAL-5 that is optimal for long data messages. Again, the call handler **116** is under the impression that it has set-up a call between the gateway interface **112** and the next exchange **118**, although in practice the call handler has, in fact, delegated the set-up to the connection supervisor which actually merely sets up the H.245 control channel. This H.245 control channel could be an AAL-5 virtual channel, an AAL-2 sub-channel within a virtual channel, or a functional equivalent. The H.245 control channel is now used to set-up the actual paths for the audio, video or data communication. These actual audio, video or data paths can use either AAL-1, AAL-2 or AAL-5. In other respects, the call set-up procedure is unaltered at a functional level, although minor and readily appreciated changes will be required to the hardware within, for example, the gateway interface **112**.

The present invention therefore advantageously provides a mechanism for interconnecting a LAN to a broadband/mini-channel network, while ostensibly maintaining conventional H.323 calls across the system. More particularly, the present invention provides an integrated architecture having increased functionality, with this accomplished without the need for significant changes in the signalling protocols of either system, other than in relation to address and port information that potentially needs to be transposed to provide inter-network addresses.

We claim:

1. A method of connecting a first network to a second network via an intermediate network, the first network and second network using a set of control messages to control media paths between the first network and the second network, the method comprising:

using a call handler independent of a switch to establish a control channel across the intermediate network to carry the set of control messages;

at a connection supervisor coupled to the switch and responsive to the call handler, intercepting the set of control messages in the intermediate network and determining a requirement for media paths, based on an interpretation of the types of intercepted control messages, in response thereto;

in response to the determination, having the connection supervisor set up media paths in the intermediate network to connect paths to carry media traffic between the first network and the second network.

2. The method of connecting according to claim **1**, wherein the set of control messages are communicated on an end-to-end basis.

3. The method of connecting according to claim **1**, wherein intercepting the control messages further includes the step of identifying the type of communication required in the media paths.

4. The method of connecting according to claim **3**, wherein the intermediate network is a broadband network.

5. The method of connecting according to claim **1**, wherein the control channel and the media paths use AAL-5.

6. The method of connecting according to claim **1**, wherein the call handler is responsive to a calling party, the method further comprises the steps of:

having the calling party dial a number of a called party;

analysing the number of the called party in the call handler and selecting an out-going route to the second network based on the number of the called party;

having the call handler instruct the connection supervisor to set-up control channel.

7. The method of connecting according to claim **1**, wherein the media paths carry at least one of audio traffic, video traffic and data traffic.

8. The method of connecting according to claim **1**, wherein the control messages are H.245 control messages.

9. The method of connecting according to claim **1**, wherein the media paths use of one AAL-1, AAL-2 and AAL-5.

10. The method of connecting according to claim **6**, further comprising having the connection supervisor indicate to the call handler that the control channel is set-up between a gateway interface and the second network.

11. The method of connecting according to claim **10**, wherein the control channel is a virtual path that used one of AAL-2 and AAL-5.

12. A method of connecting a communication traffic comprised of a plurality of traffic components across a broadband network from a local area network, the method comprising:

in the local area network, generating control messages for controlling the traffic components and applying those control messages to the broadband network;

establishing a communication path within the broadband network to carry at least one of the plurality of traffic components, the communication path established using a call handler, independent of a switch, to establish a control channel across the broadband network to carry

the control messages and wherein a connection supervisor, coupled to the switch and responsive to the call handler, intercepts the control messages to determine a requirement for media paths, based on an interpretation of the types of intercepted control messages, in response thereto, the connection supervisor setting up media paths in the broadband network to provide the communication path to carry media traffic across the broadband network from the local area network; and

in the broadband network, using the control messages to control transfer of the plurality of traffic components over the communication path.

13. The method of connecting according to claim 12, wherein the plurality of traffic components are from the set of video, audio and data traffic.

14. The method of connecting according to claim 12, wherein the communication path is a virtual channel.

15. The method of connecting according to claim 14, wherein the virtual channel comprises a plurality of mini-channels and wherein the control messages are enveloped within at least one mini-channel.

16. The method of connecting communication traffic according to claim 12, further comprising:

at the interface (112), receiving a local area network address and translating (202) said local area network address into a broadband network address.

17. The method of connecting according to claim 12, further including:

in relation to a point-to-multipoint call having a plurality of destination addresses, consolidating traffic components for each of the plurality of destination addresses into a mini-channel.

18. A connection supervisor for orchestrating the communication of traffic components between first and second networks via an intermediate network, the connection supervisor responsive, in use, to control messages communicated between the first and second networks over a control channel established by a call handler, the connection supervisor including:

means for intercepting and determining types of control messages sent across the control channel; and

means for establishing media paths dependent upon the determination of types of control messages sent across the control channel, the media paths being arranged to transfer the traffic components across the intermediate network;

wherein said connection supervisor is, in use, responsive to the call handler, the call handler being independent of a switch in the intermediate network and the connection supervisor arranged, in use, to be coupled to the switch.

19. The connection supervisor of claim 18, wherein the intermediate network is a broadband network and the communication path and the media paths are virtual channels.

20. The connection supervisor of claim 18, wherein the media paths carry at least one of audio traffic, video traffic and data traffic.

21. A communication node having a gateway that provides an interface to a first end-point in a network, the first end-point arranged to initiate a call through the communication node by sending to the gateway a called party number of a second end-point coupled to an exchange and wherein control messages are communicated between the first end-point and the second end-point, the communication node further comprising:

a call handler coupled to the gateway and responsive to the called party number, the call handler arranged to select, in response to receipt of the called party number, a control channel that supports transfer of the control messages between the gateway and the exchange, the call handler independent of a switch; and

a connection supervisor, coupled to the call handler and connectable to the switch, the connection supervisor operationally responsive to the call handler, the connection supervisor having:

(i) means for determining types of control message sent across the control channel; and

(ii) means for establishing media paths between the gateway and the exchange dependent upon the determination of types of control message sent across the control channel, the media paths being arranged to transfer traffic components across the communication node.

22. The communication node of claim 21, wherein the communication node is a broadband network and wherein the control channel and the media paths are virtual channels.

23. The communication node of claim 21, wherein the control messages are H.245 control messages.

24. The communication node of claim 21, wherein the media paths use of one AAL-1, AAL-2 and AAL-5.

* * * * *

APPENDIX D

U.S. Patent No. 6,885,658 ("Ress et al.")

US006885658B1

(12) **United States Patent**
Ress et al.

(10) **Patent No.: US 6,885,658 B1**
(45) **Date of Patent: Apr. 26, 2005**

(54) **METHOD AND APPARATUS FOR INTERWORKING BETWEEN INTERNET PROTOCOL (IP) TELEPHONY PROTOCOLS**

(75) Inventors: **David P. Ress**, Cary, NC (US); **Xuewen Li**, Cary, NC (US); **Denise J. Ingram**, Morrisville, NC (US); **Gregory Robert Utas**, Irving, TX (US)

(73) Assignee: **Nortel Networks Limited**, St. Laurent (CA)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/506,945**

(22) Filed: **Feb. 18, 2000**

Related U.S. Application Data

(60) Provisional application No. 60/137,867, filed on Jun. 7, 1999.

(51) **Int. Cl.**^{7} **H04L 12/66**; H04J 3/22

(52) **U.S. Cl.** **370/352**; 370/401; 370/466

(58) **Field of Search** 370/352–356, 370/401, 466, 351, 357, 467

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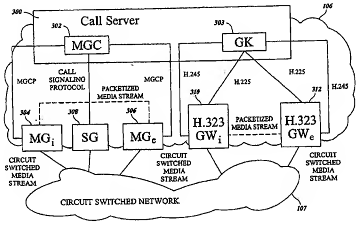
* cited by examiner

Primary Examiner—Chau T. Nguyen
Assistant Examiner—Soon D Hyun
(74) *Attorney, Agent, or Firm*—Withrow & Terranova, PLLC

(57) **ABSTRACT**

A method and an apparatus for interworking between internet protocol (IP) telephony protocols includes a call server. The call server includes a first protocol agent for communicating with a first protocol device according to a first protocol. A second protocol agent communicates with a second protocol device according to a second protocol. An interworking agent provides functions usable by the first and second protocol agents to communicate with each other according to a third protocol. The third protocol is a superset of functions provided by the first and second protocols.

34 Claims, 20 Drawing Sheets



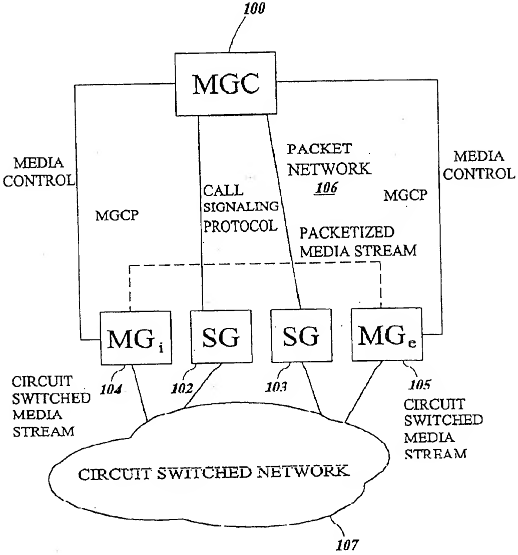


FIG. 1
(PRIOR ART)

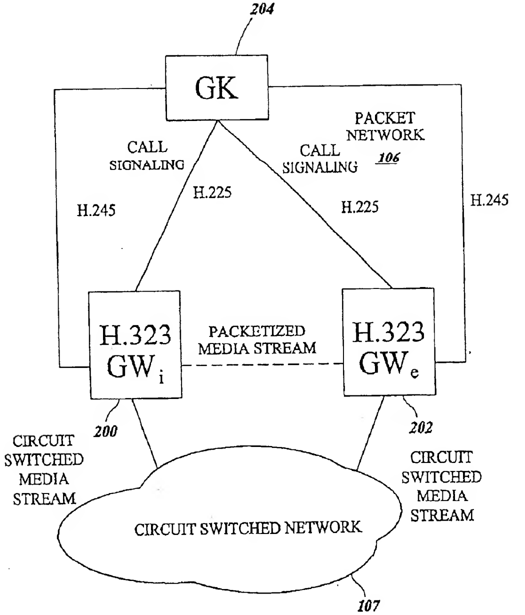


FIG. 2
(PRIOR ART)

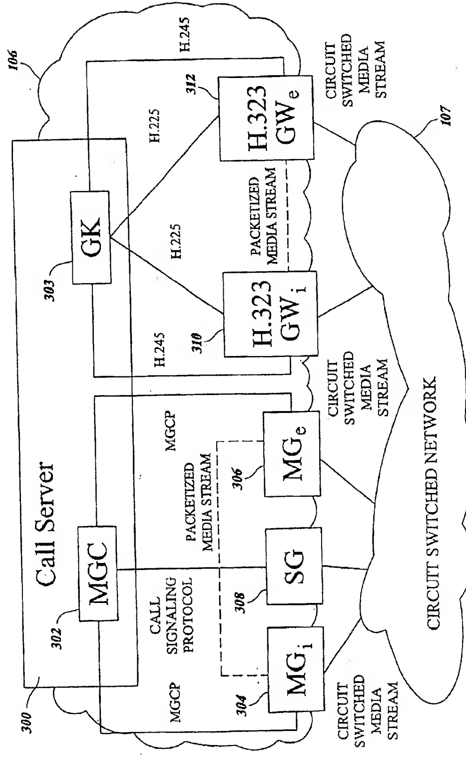

106
300
Call Server
302
MGC
303
GK
H.245
H.225
H.225
H.245
312
H.323 GWe
310
H.323 GWi
PACKETIZED MEDIA STREAM
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED MEDIA STREAM
107
MGCP
PACKETIZED MEDIA STREAM
306
MGe
CALL SIGNALING PROTOCOL
308
SG
MGCP
304
MGi
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED NETWORK

FIG. 3

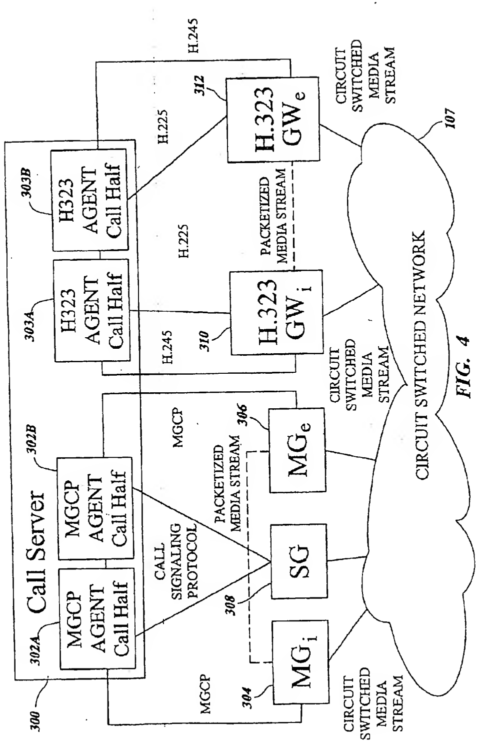

Call Server
MGCP AGENT Call Half
MGCP AGENT Call Half
H323 AGENT Call Half
H323 AGENT Call Half
300
302A
302B
303A
303B
MG_i
MG_e
SG
H.323 GW_i
H.323 GW_e
304
306
308
310
312
107
MGCP
MGCP
CALL SIGNALING PROTOCOL
PACKETIZED MEDIA STREAM
PACKETIZED MEDIA STREAM
H.245
H.245
H.225
H.225
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED NETWORK

FIG. 4

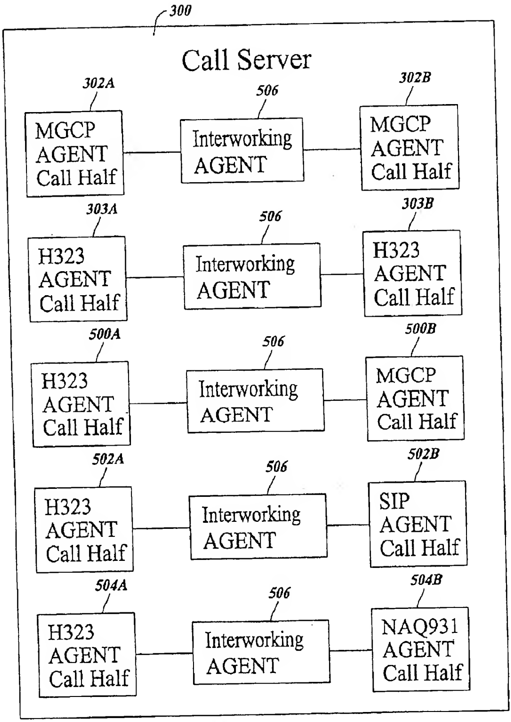

300
Call Server
302A
MGCP AGENT Call Half
506
Interworking AGENT
302B
MGCP AGENT Call Half
303A
H323 AGENT Call Half
506
Interworking AGENT
303B
H323 AGENT Call Half
500A
H323 AGENT Call Half
506
Interworking AGENT
500B
MGCP AGENT Call Half
502A
H323 AGENT Call Half
506
Interworking AGENT
502B
SIP AGENT Call Half
504A
H323 AGENT Call Half
506
Interworking AGENT
504B
NAQ931 AGENT Call Half

FIG. 5

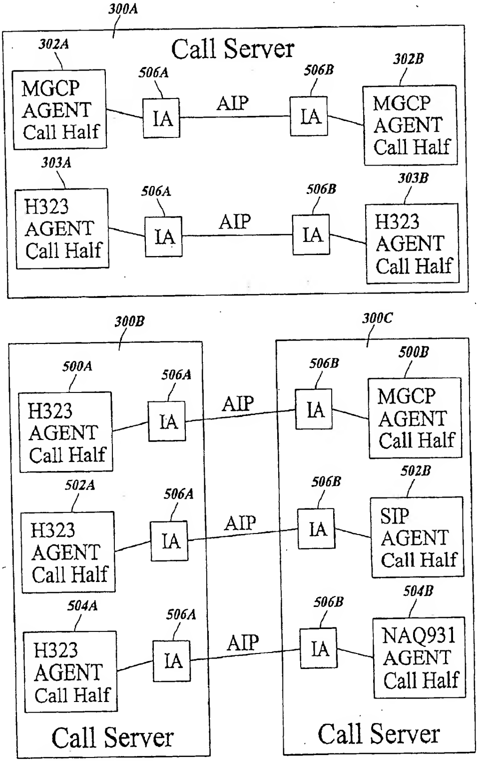

300A
Call Server
302A
MGCP
AGENT
Call Half
506A
IA
AIP
506B
IA
302B
MGCP
AGENT
Call Half
303A
H323
AGENT
Call Half
506A
IA
AIP
506B
IA
303B
H323
AGENT
Call Half
300B
500A
H323
AGENT
Call Half
506A
IA
502A
H323
AGENT
Call Half
506A
IA
504A
H323
AGENT
Call Half
506A
IA
Call Server
AIP
AIP
AIP
300C
506B
IA
500B
MGCP
AGENT
Call Half
506B
IA
502B
SIP
AGENT
Call Half
506B
IA
504B
NAQ931
AGENT
Call Half
Call Server

FIG. 6

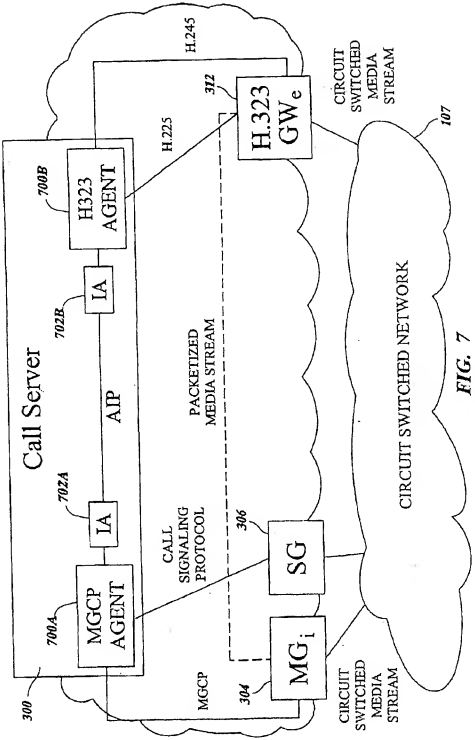

Call Server
700A
MGCP AGENT
702A
IA
AIP
702B
IA
700B
H323 AGENT
300
H.245
H.225
312
H.323 GWe
CALL SIGNALING PROTOCOL
PACKETIZED MEDIA STREAM
306
SG
304
MGi
MGCP
CIRCUIT SWITCHED MEDIA STREAM
CIRCUIT SWITCHED MEDIA STREAM
107
CIRCUIT SWITCHED NETWORK

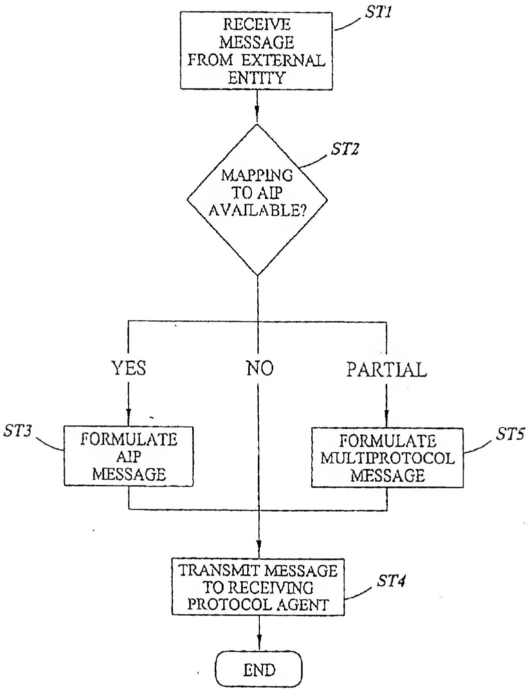
FIG. 7

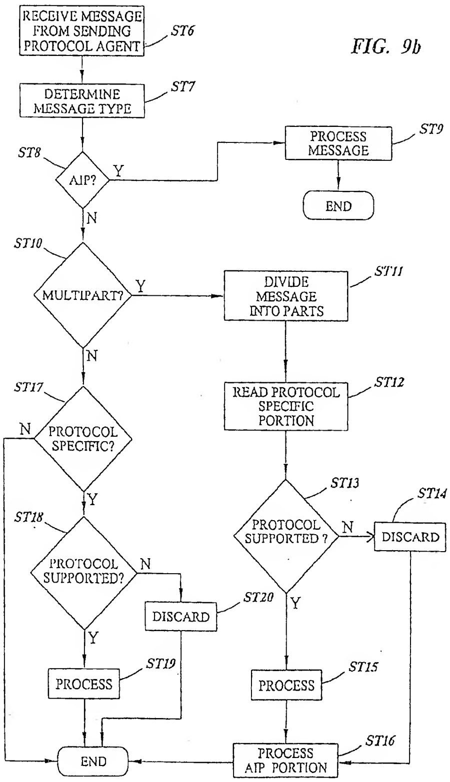
Connection Information Parameter

Field	Example Values
800 Media Type	Audio, Video, Data 802
804 Channel ID	12345 806
808 Channel Operation	No action, open, close, modify, mode change, redirect, direct, send capabilities 810
812 Current Media Description	G.711@2 frames/packet 814
816 Media Capabilities	G.711, G.729, RTP address, payload size, media specific information 818

FIG. 8

FIG. 9a


RECEIVE MESSAGE FROM EXTERNAL ENTITY
ST1
MAPPING TO AIP AVAILABLE?
ST2
YES
NO
PARTIAL
ST3
FORMULATE AIP MESSAGE
FORMULATE MULTIPROTOCOL MESSAGE
ST5
TRANSMIT MESSAGE TO RECEIVING PROTOCOL AGENT
ST4
END



RECEIVE MESSAGE FROM SENDING PROTOCOL AGENT
ST6
DETERMINE MESSAGE TYPE
ST7
ST8
AIP?
Y
N
PROCESS MESSAGE
ST9
END
ST10
MULTIPART?
Y
N
DIVIDE MESSAGE INTO PARTS
ST11
READ PROTOCOL SPECIFIC PORTION
ST12
ST17
PROTOCOL SPECIFIC?
N
Y
ST18
PROTOCOL SUPPORTED?
N
Y
DISCARD
ST20
PROCESS
ST19
END
ST13
PROTOCOL SUPPORTED ?
N
Y
DISCARD
ST14
PROCESS
ST15
PROCESS AIP PORTION
ST16

FIG. 9b

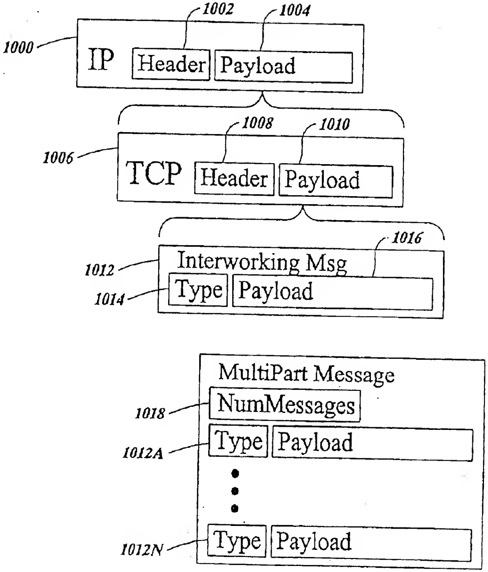

1002
1004
1000
IP
Header
Payload
1008
1010
1006
TCP
Header
Payload
1016
1012
Interworking Msg
1014
Type
Payload
MultiPart Message
1018
NumMessages
1012A
Type
Payload
1012N
Type
Payload

FIG. 10

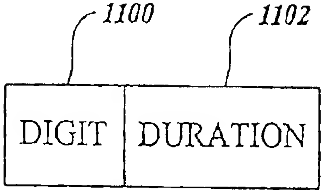


FIG. 11

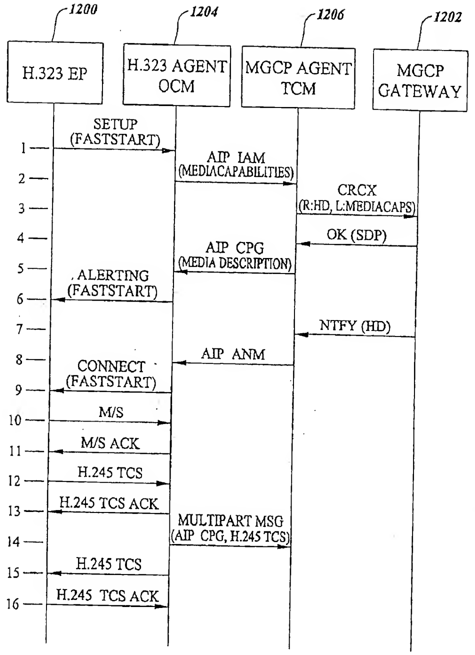

1200
1204
1206
1202
H.323 EP
H.323 AGENT OCM
MGCP AGENT TCM
MGCP GATEWAY
SETUP (FASTSTART)
AIP IAM (MEDIACAPABILITIES)
CRCX (R:HD, L:MEDIACAPS)
OK (SDP)
AIP CPG (MEDIA DESCRIPTION)
ALERTING (FASTSTART)
NTFY (HD)
AIP ANM
CONNECT (FASTSTART)
M/S
M/S ACK
H.245 TCS
H.245 TCS ACK
MULTIPART MSG (AIP CPG, H.245 TCS)
H.245 TCS
H.245 TCS ACK
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16

FIG. 12

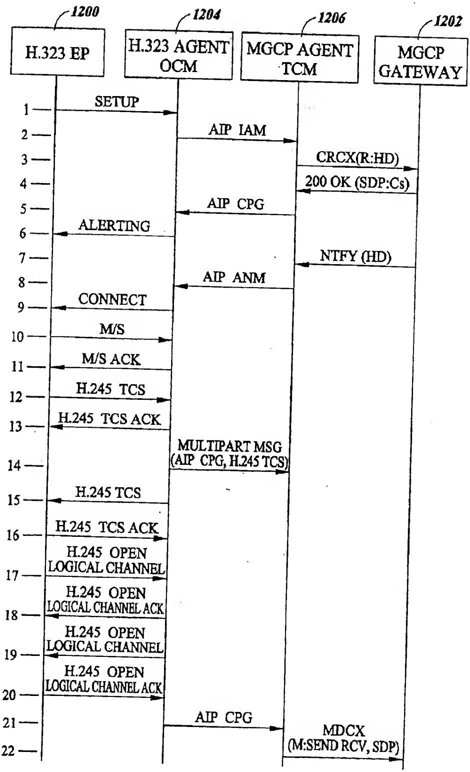

1200
1204
1206
1202
H.323 EP
H.323 AGENT OCM
MGCP AGENT TCM
MGCP GATEWAY
1
SETUP
2
AIP IAM
3
CRCX(R:HD)
4
200 OK (SDP:Cs)
5
AIP CPG
6
ALERTING
7
NTFY (HD)
8
AIP ANM
9
CONNECT
10
M/S
11
M/S ACK
12
H.245 TCS
13
H.245 TCS ACK
14
MULTIPART MSG (AIP CPG, H.245 TCS)
15
H.245 TCS
16
H.245 TCS ACK
17
H.245 OPEN LOGICAL CHANNEL
18
H.245 OPEN LOGICAL CHANNEL ACK
19
H.245 OPEN LOGICAL CHANNEL
20
H.245 OPEN LOGICAL CHANNEL ACK
21
AIP CPG
22
MDCX (M:SEND RCV, SDP)

FIG. 13

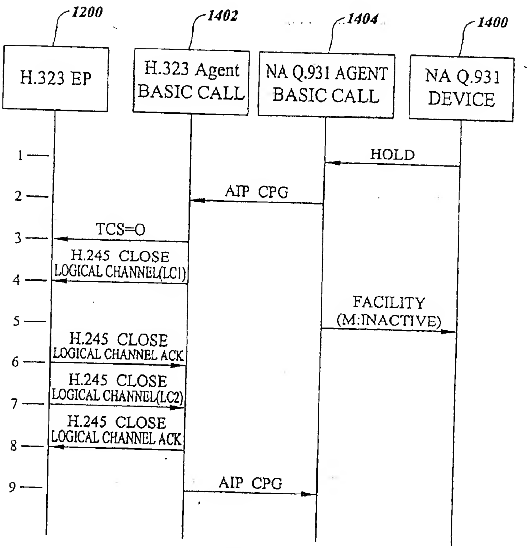

1200
1402
1404
1400
H.323 EP
H.323 Agent BASIC CALL
NA Q.931 AGENT BASIC CALL
NA Q.931 DEVICE
1
HOLD
2
AIP CPG
3
TCS=O
H.245 CLOSE LOGICAL CHANNEL(LC1)
4
5
FACILITY (M:INACTIVE)
H.245 CLOSE LOGICAL CHANNEL ACK
6
H.245 CLOSE LOGICAL CHANNEL(LC2)
7
H.245 CLOSE LOGICAL CHANNEL ACK
8
9
AIP CPG

FIG. 14

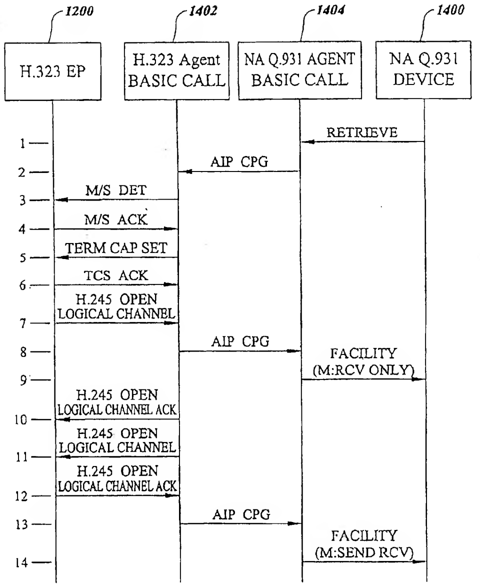

1200
1402
1404
1400
H.323 EP
H.323 Agent BASIC CALL
NA Q.931 AGENT BASIC CALL
NA Q.931 DEVICE
RETRIEVE
AIP CPG
M/S DET
M/S ACK
TERM CAP SET
TCS ACK
H.245 OPEN LOGICAL CHANNEL
AIP CPG
FACILITY (M:RCV ONLY)
H.245 OPEN LOGICAL CHANNEL ACK
H.245 OPEN LOGICAL CHANNEL
H.245 OPEN LOGICAL CHANNEL ACK
AIP CPG
FACILITY (M:SEND RCV)
1
2
3
4
5
6
7
8
9
10
11
12
13
14

FIG. 15

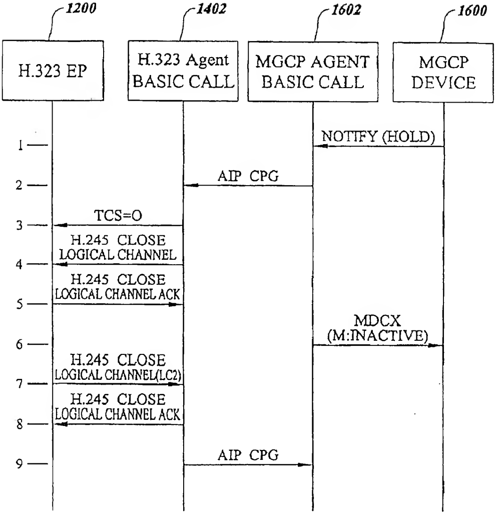

1200
1402
1602
1600
H.323 EP
H.323 Agent BASIC CALL
MGCP AGENT BASIC CALL
MGCP DEVICE
1
2
3
4
5
6
7
8
9
NOTIFY (HOLD)
AIP CPG
TCS=O
H.245 CLOSE LOGICAL CHANNEL
H.245 CLOSE LOGICAL CHANNEL ACK
MDCX (M:INACTIVE)
H.245 CLOSE LOGICAL CHANNEL(LC2)
H.245 CLOSE LOGICAL CHANNEL ACK
AIP CPG

FIG. 16

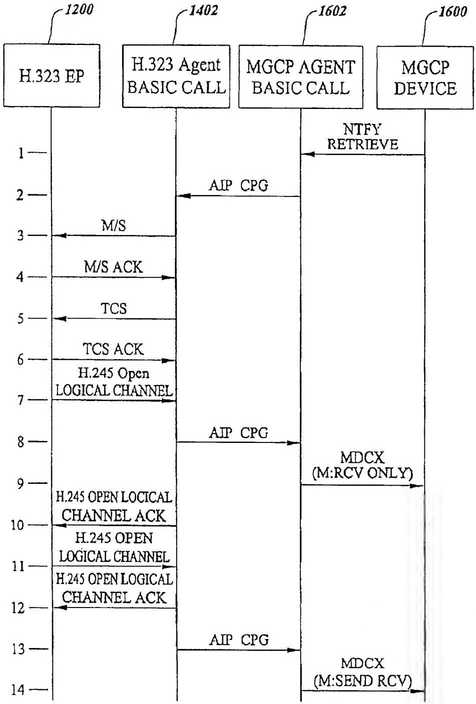
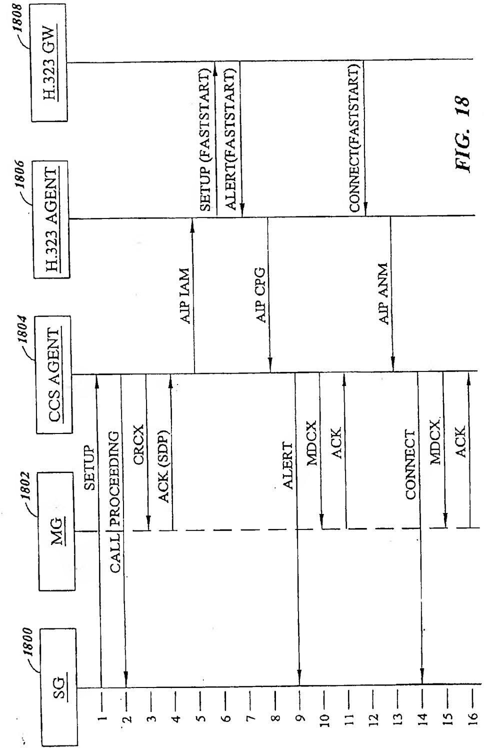

1200
1402
1602
1600
H.323 EP
H.323 Agent BASIC CALL
MGCP AGENT BASIC CALL
MGCP DEVICE
NTFY RETRIEVE
AIP CPG
M/S
M/S ACK
TCS
TCS ACK
H.245 Open LOGICAL CHANNEL
AIP CPG
MDCX (M:RCV ONLY)
H.245 OPEN LOGICAL CHANNEL ACK
H.245 OPEN LOGICAL CHANNEL
H.245 OPEN LOGICAL CHANNEL ACK
AIP CPG
MDCX (M:SEND RCV)
1
2
3
4
5
6
7
8
9
10
11
12
13
14

FIG. 17



1800
SG
1802
MG
1804
CCS AGENT
1806
H.323 AGENT
1808
H.323 GW
SETUP
CALL PROCEEDING
CRCX
ACK (SDP)
AIP IAM
SETUP (FASTSTART)
ALERT(FASTSTART)
AIP CPG
ALERT
MDCX
ACK
CONNECT(FASTSTART)
AIP ANM
CONNECT
MDCX
ACK
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16

FIG. 18

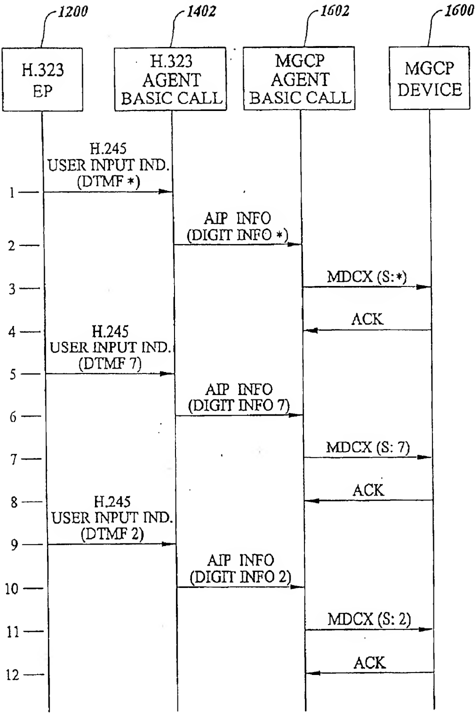

1200
1402
1602
1600
H.323
EP
H.323
AGENT
BASIC CALL
MGCP
AGENT
BASIC CALL
MGCP
DEVICE
H.245
USER INPUT IND.
(DTMF *)
1
AIP INFO
(DIGIT INFO *)
2
MDCX (S:*)
3
ACK
4
H.245
USER INPUT IND.
(DTMF 7)
5
AIP INFO
(DIGIT INFO 7)
6
MDCX (S: 7)
7
ACK
8
H.245
USER INPUT IND.
(DTMF 2)
9
AIP INFO
(DIGIT INFO 2)
10
MDCX (S: 2)
11
ACK
12

FIG. 19

METHOD AND APPARATUS FOR INTERWORKING BETWEEN INTERNET PROTOCOL (IP) TELEPHONY PROTOCOLS

PRIORITY APPLICATION

This Application claims the benefit of U.S. Provisional Patent Application Ser. No. 60/137,867 filed Jun. 7, 1999, the disclosure of which is incorporated herein by reference in its entirety.

TECHNICAL FIELD

The present invention relates to a method and an apparatus for interworking between communications protocols. More particularly, the present invention relates to a method and an apparatus for interworking between internet protocol (IP) telephony protocols.

BACKGROUND ART

There are a variety of known protocols for establishing media stream communications, such as voice, data, video, or combinations thereof, over an IP network. Protocols for establishing media stream communications over an IP network are referred to herein as IP telephony protocols. One example of an IP telephony protocol is the media gateway control protocol (MGCP). MGCP defines signals and events by which a software entity, known as a media gateway (MG), is controlled by another software entity, known as a media gateway controller (MGC), in a packet network. The media gateway controller processes call control signaling from one or more signaling gateways (SGs) and utilizes MGCP media control signaling to establish media streams between MGs. An MGC that processes call control signaling in this manner is also referred to as a call agent. The terms media gateway controller and call agent are used interchangeably herein. The media gateway controller performs call control functions, such as translations, resource management, media capabilities negotiation and selection, and media stream management. It can also provide additional services.

FIG. 1 illustrates conventional communications using MGCP. In FIG. 1, MGC **100** receives call signaling information from SGs **102** and **103** and controls MGs **104** and **105** to establish packetized media stream communications between end users in packet network **106**. For example, SG **102** and MG **104** can be associated with a calling party end user device for a given media stream communication. Similarly, SG **103** and MG **105** can be associated with a called party end user device for a given media stream communication. MGC **100** can control MGs **104** and **105** to establish media stream communications between the called and calling end user devices, such as PSTN terminals.

A detailed explanation of MGCP is found in Media Gateway Control Protocol (MGCP), Version 0.1 Draft, Internet Engineering Task Force, Feb. 21, 1999, the disclosure of which is incorporated herein by reference in its entirety.

Another example of an IP telephony protocol is International Telecommunications Union (ITU) Recommendation H.323. H.323 defines a protocol by which endpoints, such as gateways, terminals, or multipoint control units MCUs), can place calls in a packet network. A gateway translates between circuit-switched and packet-switched communication protocols. A terminal is a device, such as an IP terminal, that provides end user access to a network. An MCU is a device that supports conferences between three or more endpoints. H.323 defines a gatekeeper as an entity that provides address translation and controls access to the packet network for H.323 endpoints. The gatekeeper can also provide additional services, such as call control and supplementary services.

FIG. 2 illustrates an example of conventional H.323 communications. In FIG. 2, a first gateway **200** can be associated with a calling end user device and a second gateway **202** can be associated with a called end user device for a given media communication. Gatekeeper **204** performs call signaling functions, such as call setup and teardown, to establish calls between end user devices associated with gateways **200** and **202**. The end user devices can be PTSN terminals connected to gateway **200**. Alternatively, gateway **200** can be omitted and replaced by H.323 terminals or H.323 MCUs. Once gatekeeper **204** performs the call signaling functions necessary to set up a call, the media stream for the call flows between gateways **200** and **202**. Detailed information relating to H.323 can be found in ITU Recommendation H.323, Packet Based Multimedia Communications Systems, February 1998, the disclosure of which is incorporated herein by reference in its entirety.

Yet another IP telephony protocol is ITU Recommendation H.248. The Internet Engineering Task Force (IETF) formed the MEGACO Group to evolve the MGCP protocol. As the MEGACO Group matured the protocol, the MEGACO Group allied itself with the ITU, and the specification developed by the MEGACO Group has become known as ITU Recommendation H.248. Thus, ITU recommendation H.248 can be viewed similarly to MGCP.

Another IP telephony protocol is the session initiation protocol (SIP). SIP is an application layer signaling protocol for creating, modifying, and terminating sessions between one or more participants. The sessions include internet multimedia conferences, internet telephone calls, and multimedia distribution. SIP originated from Columbia University and is gaining acceptance as a protocol for exchanging call signaling information over a packet network. A detailed description of SIP can be found in Request for Comments (RFC) 2543 SIP: Session Initiation Protocol, March 1999, the disclosure of which is incorporated herein by reference in its entirety.

In addition to the published protocols described above, many vendors of telecommunications equipment and services are supporting IP telephony applications via proprietary protocols.

All of the IP telephony protocols described above are being implemented by various vendors. However, standards for interworking equipment that communicates using one protocol with equipment that communicates using another protocol are immature, nonexistent, or focus only on a specific type of application. Accordingly, there exists a long-felt need for a novel method and apparatus for interworking between IP telephony protocols.

DISCLOSURE OF THE INVENTION

The present invention provides a novel method and apparatus for interworking between IP telephony protocols. Although most of the examples described below relate to MGCP and H.323, it is understood that the method and apparatus described herein are applicable to any IP telephony protocol.

Many of the protocols described herein define an entity that is responsible for performing functions and requests on behalf of a telephony device. Typically, these functions and requests include translations, media capabilities exchange, and other services. The entities that perform the functions

can be logical, physical, or both. For example, in MGCP, the MGC or call agent performs call signaling functions on behalf of a gateway. In H.323, the gatekeeper performs call signaling functions for an H.323 gateway. In SIP, a proxy server performs call signaling functions for an end user. In order to facilitate a description of the present invention, the term call server is used herein to refer to an entity that performs call signaling functions, such as translations and media capabilities exchange, on behalf of an end user device, gateway, or other entity.

According to a first aspect, the present invention includes a call server including a first protocol agent and a second protocol agent. The first protocol agent communicates with a first protocol device according to a first protocol. The second protocol agent communicates with a second protocol device according to a second protocol. An interworking agent provides functions usable by the first and second protocol agents to communicate using a third protocol. The third protocol provides a superset of the functions provided by the first and second protocols.

Accordingly, it is an object of the present invention to provide a novel method and apparatus for interworking between IP telephony protocols.

An object of the invention having been stated hereinabove, and which is achieved in whole or in part by the present invention, other objects will be evident as the description proceeds, when taken in connection with the accompanying drawings as best described hereinbelow.

BRIEF DESCRIPTION OF THE DRAWINGS

A description of the present invention will now proceed with reference to the accompanying drawing of which:

FIG. **1** is a block diagram illustrating conventional MGCP network entities;

FIG. **2** is a block diagram illustrating conventional H.323 network entities;

FIG. **3** is a block diagram illustrating media gateway controller and gatekeeper functions implemented within a call server according to an embodiment of the present invention;

FIG. **4** is a block diagram illustrating a call server wherein each call half is represented by an agent according to an embodiment of the present invention;

FIG. **5** is a block diagram illustrating a call server including a plurality of interworking agents according to an embodiment of the present invention;

FIG. **6** is a block diagram illustrating protocol agents implementing originating and terminating call half functions executing on different machines wherein an interworking agent is associated with each protocol agent;

FIG. **7** is a block diagram illustrating a call server including MGCP, interworking, and H.323 agents for interworking MGCP and H.323 entities according to an embodiment of the present invention;

FIG. **8** is a block diagram illustrating a connection information parameter data structure according to an embodiment of the present invention;

FIGS. 9(*a*) and 9(*b*) are flow charts illustrating message tunneling according to an embodiment of the present invention;

FIG. **10** is a block diagram illustrating exemplary agent interworking protocol message structures according to an embodiment of the present invention;

FIG. **11** is a block diagram illustrating an exemplary data structure for a digit information parameter according to an embodiment of the present invention;

FIG. **12** is a call flow diagram illustrating exemplary call signaling for H.323 fast start to MGCP communications according to an embodiment of the present invention;

FIG. **13** is a call flow diagram illustrating H.323 non-fast-start to MGCP communications according to an embodiment of the present invention;

FIG. **14** is a call flow diagram illustrating exemplary call signaling for a hold scenario between H.323 and North American Q.931 endpoints according to an embodiment of the present invention;

FIG. **15** is a call flow diagram illustrating exemplary call signaling for a retrieve scenario between H.323 and North American Q.931 endpoints according to an embodiment of the present invention;

FIG. **16** is a call flow diagram illustrating exemplary call signaling for a hold scenario between H.323 and MGCP endpoints according to an embodiment of the present invention;

FIG. **17** is a call flow diagram illustrating exemplary call signaling for a retrieve scenario between H.323 and MGCP endpoints according to an embodiment of the present invention;

FIG. **18** is a call flow diagram illustrating exemplary call signaling between H.323 and MGCP endpoints for common channel signaling according to an embodiment of the present invention; and

FIG. **19** is a call flow diagram illustrating exemplary call signaling between an MGCP gateway and an H.323 gateway according to an embodiment of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention provides a novel method and apparatus for interworking between IP telephony protocols. In order to provide this interworking, a call server includes agents that communicate with other entities according to the protocols implemented by the other entities. However, the protocol agents communicate with each other utilizing a protocol-independent agent interworking protocol (AIP). As a result, network entities that implement different protocols can seamlessly communicate with each other.

FIG. **3** illustrates a call server including MGC and GK functions according to an embodiment of the present invention. In FIG. **3**, a call server **300** includes an MGC function **302** and a GK function **303**. The call server is a software entity that can execute on a single machine or on multiple machines. MGs and SGs recognize call server **300** as an MGC. H.323 endpoints, such as gateways, recognize call server **300** as a GK. For example, in the illustrated embodiment, ingress MG **304**, egress MG **306**, and SG **308** recognize call server **300** as an MGC. Similarly, ingress H.323 gateway **310** and egress H.323 gateway **312** recognize call server **300** as a gatekeeper.

In order for MGs **304** and **306** to recognize call server **300** as an MGC, MGC function **302** in call server **300** is adapted to communicate with MGs **304** and **306** using MGCP. Similarly, in order for SG **308** to recognize call server **300** as an MGC, MGC function **302** in call server **300** communicates with SG **308** using a call signaling protocol, such as ISDN Part (ISUP). In order for H.323 gateways **310** and **312** to recognize call server **300** as a gatekeeper, gatekeeper function **303** in call server **300** communicates with gateways **310** and **312** according to ITU Recommendations H.225 and H.245.

FIG. **4** illustrates an embodiment of call server **300** in which the call processing functions illustrated in FIG. **3** are

separated into call agents, each of which performs call half functions. As used herein, call half functions refer to functions associated with either the originating or terminating side of a call. For example, in FIG. 4, MGCP function **302** illustrated in FIG. 3 is divided into MGCP agent **302A** and MGCP agent **302B**. Similarly, H.323 function **303** illustrated in FIG. 3 is divided into H.323 agent **303A** and H.323 agent **303B**. MGCP agent **302A** and H.323 agent **303A** can perform call originating functions, such as collection of digits and translations. MGCP agent **302B** and H.323 agent **303B** can perform call terminating functions, such as trunk selection and alerting the called party of an incoming call. The functions performed by each call half will be explained in detail below with reference to call flow diagrams.

As mentioned above, the present invention is not limited to interworking between MGCP and H.323 entities. For example, FIG. **5** illustrates a call server including protocol agents configured to communicate with other agents using a variety of different protocols. In the illustrated embodiment, call server **300** includes MGCP agents **302A** and **302B** for processing MGCP to MGCP calls, H.323 agents **303A** and **303B** for processing H.323 to H323 calls, H.323 agent **500A** and MGCP agent **500B** for processing H.323 to MGCP calls, H.323 agent **502A** and SIP agent **502B** for processing H.323 to SIP calls, and H.323 agent **504A** and NAQ.931 agents **504B** for processing H.323 to NAQ.**931** calls. In addition, to the protocol agents, call server **300** also includes an interworking agent **506** which facilitates communication between protocol agents. More particularly, interworking agent **506** includes methods for getting and setting AIP parameters, building AIP messages, and establishing and maintaining connections, such as TCP or reliable UDP connections, between protocol agents. Interworking agents can also identify AIP message types, which will be described in more detail below. Thus, as illustrated in FIG. **5**, interworking agent **506** provides functions usable by a variety of different protocol agents to provide seamless interworking between the protocol agents.

In a preferred embodiment of the invention, the interworking agent is divided into separate software components, one component associated with the protocol agent for each call half. The division of the interworking agent into two software components allows protocol agents associated with a given call to execute on separate machines.

Referring to FIG. 6, call server **300** illustrated in FIG. **5** is divided into call servers **300A**, **300B**, and **300C**, which can execute on the same machine or on different machines. Call server **300A** includes protocol agents that perform both originating and terminating call half functions. Call servers **300B** and **300C** each include protocol agents that perform only originating or terminating call half functions. This division of call processing functionality is enabled by interworking agents components **506A** and **506B**, which enable protocol agents to communicate with each other using AIP messaging. Exemplary information that can be exchanged using AIP messaging includes information regarding call progress, media capabilities and addresses, supplementary services, etc. By allowing the protocol agents to reside on separate machines, the interworking agents according to embodiments of the present invention allow efficient division of call processing functions.

FIG. 7 illustrates an example of an MGCP-H.323 network typology wherein communication between MGCP and H.323 endpoints occurs through a call server according to an embodiment of the present invention. In FIG. **7**, call server **300** includes MGCP agent **700A** for performing originating call half functions according to the media gateway control protocol and H.323 agent **700B** for performing terminating call half functions according to the H.323 protocol. More particularly, MGCP agent **700A** communicates with ingress media gateway **304** according to the media gateway control protocol and with signaling gateway **306** according to a call signaling protocol, such as ISUP. H.323 agent **700B** communicates with H.323 gateway **312** according to H.225 and H.245 protocols. MGCP agent **700A** and H.323 agent **700B** communicate with each other using AIP messaging. Interworking agent components **702** and **702B** provide the functions that protocol agents **700A** and **700B** use to formulate and process AIP messages.

Because the interworking agent components **702A** and **702B** provide functions for converting messages to and from a protocol independent format, MGCP agent **700A** and the H.323 agent **700B** need not be aware of each other's protocol. Similarly, MG **304** and SG **306** need not be aware of the protocol of H.323 gateway **312**, and H.323 gateway **312** need not be aware of the protocol of MG **304** and SG **306**.

Agent Interworking Protocol

As stated above, interworking agents according to embodiments of the present invention communicate with each other according to a protocol independent format referred to as the agent interworking protocol. The agent interworking protocol is preferably capable of representing a reasonable superset of the messaging capabilities of all protocols to be supported within the packet network. Designing an interworking protocol that supports all of the capabilities of all of the supported protocols is an unnecessarily burdensome task since some capabilities are rarely used or are only useful when communicating between devices that support the particular protocol. In addition, these rarely used capabilities can be communicated between agents that support these capabilities using tunneling, as will be described in more detail below. Accordingly, it is desirable that the agent interworking protocol provide a reasonable superset of the capabilities of supported protocols.

Rather than designing an entirely new protocol for use as the agent interworking protocol, it is more desirable to select an existing protocol that comes close to meeting the superset definition described above and extending that protocol. Existing protocols that could be used as the base protocol for the agent interworking protocol include Q.931, ISUP, and SIP. The agent interworking protocol implemented in interworking agents according to preferred embodiments of the present invention is based upon ISUP. For example, AIP includes traditional ISUP messages such as initial address messages (IAM), answer messages (ANM), and release messages (REL). The agent interworking protocol extends the base protocol to include additional procedures and signaling required to meet interoperability requirements. The functions and data structures used in the agent interworking protocol to meet these requirements will now be discussed in more detail.

One function that must be provided by the agent interworking protocol is a method for exchanging media capabilities between protocol agents. Each of the agent protocols to be interworked provide some means by which a telephony device can make known the media capabilities that it supports. These capabilities must be exchanged between two devices that desire to participate in a media stream communication in order to select a mutually compatible media session definition.

Capabilities Exchange Between H.323 Devices

H.323 allows an endpoint to advertise its capabilities at two different times—during call establishment and after call

establishment. For example, some H.323 devices support fast start capabilities which allow a partial list of media capabilities to be exchanged in H.225 call establishment messages. This method of exchanging capabilities allows faster establishment of a media stream between endpoints because capabilities are exchanged during call signaling, rather than waiting until after call signaling has been completed. In order to exchange capabilities after call establishment, H.323 compliant devices use H.245 signaling to provide a full description of all media capabilities supported.

Capabilities Exchange Between MGCP and SIP Devices

MGCP and SIP support the use of the session description protocol (SDP) for encoding the capabilities supported by the device. The session description protocol is included in call establishment messages similarly to H.323 fast start messages. For example, a SIP call establishment message, such as an INVITE message, includes an SDP portion in the body of the message. The SDP portion includes the capabilities supported by the endpoint, such as encoding and decoding algorithms, type of media stream, etc.

Because these capabilities can be exchanged during call setup or after call establishment, the agent interworking protocol implemented in call servers according to embodiments of the present invention is preferably flexible enough to support capabilities exchange at either time. In addition, because each of the above-mentioned protocols uses different syntax for specifying the capability's definition, AIP preferably provides a normalized syntax to which interworking agents can map the capability's definitions.

Media Management

In addition to providing a method for exchanging media capabilities, the agent interworking protocol preferably also provides media management capabilities that include a reasonable superset of the media management capabilities of supported protocols. For example, each of the agent protocols provide support for establishing and altering media streams; however, the specific protocols vary significantly. H.323 fast start procedures allow H.323 devices to establish a media stream in concert with call establishment. However, fast start is optional and might not be supported by a given H.323 device. H.245 procedures allow H.323 devices to open and close media channels post call establishment. H.323 is very limited in its ability to alter a media stream once established. H.323 media streams can be unidirectional or bi-directional. Voice/audio media is typically represented via two independent unidirectional streams on IP networks with bi-directional media being typically used for data or for voice on ATM networks.

MGCP supports establishment of media streams during call establishment similar to H.323 fast start procedures. Media streams can be either unidirectional or bi-directional and can be changed from one format to the other at any time during a call. MGCP allows media channels to be modified in a variety of ways without having to be closed. For example, a media stream can be redirected by changing the receiving real time protocol (RTP) address. The encoding format can be changed by changing the codec. The mode can be changed to send only, send receive, receive only, or inactive. SIP is similar to MGCP in its ability to modify media streams.

In order to provide an interworking solution that accommodates these agent protocols, three design objectives are preferably met. The first design objective is that the agent interworking protocols must provide sufficient flexibility to meet the requirements of all agent protocols. Second, the agent design preferably maps between agent specific and AIP procedures and syntax for media management. The third objective is that a flexible control framework is preferably implemented that allows the agent to easily react to media changes made by the agent implementing the other call half. The connection information parameter illustrated in FIG. **8** is the mechanism provided by the agent interworking protocol for implementing media management functions and exchanging media capabilities. The times for exchanging media capabilities and performing media management functions according to the agent interworking protocol will be described in more detail below with respect to the call flow diagrams.

FIG. **8** is a table illustrating exemplary fields and field values for the connection information parameter according to an embodiment of the present invention. In FIG. **8**, the left hand column represents the fields in the connection information parameter data structure. The right hand column represents example values for each of the fields in the left hand column. In the illustrated embodiment, the connection information parameter includes a media type field **800** that holds a media type value **802** for specifying the type of media being exchanged or sought to be exchanged in a media stream. Example values for the media type field include audio, video, and data. Channel ID field **804** includes an internally assigned channel ID value that allows an interworking agent to identify the media stream. In the illustrated embodiment, **12345** is given as an example value **806** for channel ID field **804**. Channel operation field **808** includes a channel operation value **810** for specifying the operation being performed on the media stream. Values **810** for the channel operation field **808** are preferably a superset of protocol media stream operations for the supported protocols. In the illustrated embodiment, exemplary values for the channel operation field are no action, open, close, modify, mode change, redirect, direct, and send capabilities. The no action value indicates that no change is being made to the existing media stream. The open value specifies that a media stream is sought to be opened. The close value indicates that an open media stream is sought to be closed. The modify value indicates that the media stream is sought to be modified, e.g., changing a codec from G.711 to G.729a. The mode change value indicates that the mode of the media stream is sought to be changed, e.g., from send only to receive only. The redirect value indicates that the media stream is to be redirected to another endpoint. The direct value specifies the location to which the media stream is to be directed. The send capabilities value requests the receiving entity to transfer the media capabilities list.

Current media description field **812** stores current media description value **814** for indicating the description of the current media stream. In the illustrated embodiment, an example of a current media description value is G.711 at two frames per packet. Media capabilities field **816** includes media capabilities value or values **818** that allows an entity to exchange its media capabilities with another entity. In the illustrated example, the media capabilities field includes a list of supported formats, such as G.711, G.729. Media capabilities field **818** also includes a payload size value that specifies the size of media capabilities field. Media capabilities field **818** also includes a redefinable area in which information specific to the type of media and codes can be specified. For example, a facsimile media stream requires certain attributes that are not required for other media types. The redefinable area allows this information to be specified.

Message Tunneling

As described above, the agent interworking protocol represents a reasonable superset of the agent protocols sufficient to achieve interworking. However, the agent interworking protocol is not a complete superset of the supported protocols. That is, certain agent protocols can contain messages or parameters which do not map to any other agent protocols, but provide added value for a call between two devices of the same type. In this case, the agent interworking protocol preferably supports tunneling of the native protocol message. As used herein, tunneling refers to transferring the native protocol message from one protocol agent to another protocol agent without converting to and from the agent interworking protocol. The agent receiving the native protocol message can inspect the message, and if the agent understands the message, process the message accordingly.

An example of when it might be desirable to tunnel a message relates to H.323. H.323 provides a sophisticated means of representing terminal capabilities in sets. The agent interworking protocol, as described herein, might not include functionality for representing terminal capabilities in sets as defined in H.323, because the other protocols do not support such capability. Another capability that H.323 supports which can not be supported by other protocols is the exchange of H.245 indications between two H.323 devices. Some of these indications have no equivalent mapping to other agent protocols. In these situations, it can be desirable to tunnel the H.323 messages from one agent to another agent.

Method of Tunneling Messages

According to an embodiment of the present invention, interworking messages can be of three types:

- Agent Interworking protocol messages—protocol-neutral messages understood by all protocol agents;
- Native protocol messages—protocol-specific messages, such as SIP, MGCP, and H.323 messages;
- Multipart messages—messages that contain multiple other messages, such as native protocol messages and AIP messages. All agents are preferably capable of extracting the AIP message and processing the message accordingly. If the multipart message contains a native protocol message, this message is preferably processed if supported.

FIGS. **9**(*a*) and **9**(*b*) are flow charts illustrating exemplary formulating and processing of interworking messages by a call server according to an embodiment of the present invention. The flow chart in FIG. **9**(*a*) illustrates exemplary steps that can be performed by a sending protocol agent in formulating an interworking message using procedures provided by an associated interworking agent. The flow chart in FIG. **9**(*b*) illustrates exemplary steps that can be performed by a receiving protocol agent using procedures provided by an associated interworking agent upon receiving an interworking message. Referring to FIG. **9**(*a*), in step **ST1**, the sending protocol agent receives a message from an external entity, such as an H.323 gateway. In step **ST2**, the sending protocol agent determines whether a mapping is available to the agent interworking protocol. In step **ST3**, if a mapping is available, the sending protocol agent formulates the corresponding AIP message using functions provided by the interworking agent associated with the sending protocol agent (hereinafter, "the first interworking agent") and transmits the message to the receiving protocol agent (step ST4). In step **ST2**, if the sending protocol agent determines that the mapping to the agent interworking protocol is not available, the sending protocol agent simply transmits the protocol message without modification to the receiving protocol agent (step ST4). In step **ST2**, if the sending protocol agent determines that a mapping to AIP is partially available, the sending protocol agent can formulate a multiprotocol message including the AIP message and the native protocol message (step **ST5**). The sending protocol agent can then transmit the multiprotocol message to the receiving protocol agent (step ST4).

Referring to FIG. **9**(*b*), in step **ST6**, the receiving protocol agent receives the message from the sending protocol agent. In step **ST7**, receiving protocol agent determines the message type, i.e., whether the message is a protocol specific message, an agent interworking protocol message, or a multipart message, using procedures provided by its associated interworking agent (hereinafter, "the second interworking agent").

In step **ST8**, if the receiving protocol agent determines that the message is an agent interworking protocol message, the receiving protocol agent processes the message (step **ST9**). In step **ST10**, the receiving protocol agent determines whether the message is a multipart message. If the message is a multipart message, the receiving protocol agent separates the multi-protocol message into its component messages (step **ST11**). After the receiving protocol agent separates the message, the receiving protocol agent reads the protocol specific portion of the AIP message (step **ST12**). In step **ST13**, the receiving protocol agent determines whether the protocol in the protocol specific portion is supported. If the protocol is not supported, the receiving protocol agent discards the message (step **ST14**). If the protocol is supported, the receiving protocol agent processes the message (step **ST15**). In step **ST16**, the receiving protocol agent processes the AIP portion of the message.

Referring to step **ST17**, if the receiving protocol agent determines that the message is a protocol specific message, the receiving protocol agent determines whether the protocol of the message is supported (step **ST18**). If the protocol is supported, the receiving protocol agent processes the message (step **ST19**). If the protocol is not supported, the receiving protocol agent discards the message (step **ST20**).

Transport Mechanism for Agent Interworking Messages

Interworking messages can be sent between protocol agents using any packet based protocol, for example, TCP, UDP, etc. In a preferred embodiment of the invention, interworking messages are transmitted between interworking agents using TCP over IP. As is known in the art, an IP message includes a header portion and a data portion. A TCP message is encapsulated in the data portion of the IP message. The TCP message also includes a header portion and a data portion. Interworking messages are encapsulated in the data portion of the TCP message. Interworking messages also include a header portion and a data portion. The header portion indicates the message type, i.e., AIP, protocol-specific, or multipart.

FIG. **10** illustrates the relationships between IP messages, TCP messages, and interworking messages. In FIG. **10**, IP message **1000** includes a header portion **1002** and a data portion **1004**. TCP message **1006** is encapsulated in data portion **1004** of IP message **1000**. TCP message **1006** includes a header portion **1008** and a data portion **1010**.

Interworking message **1012** is encapsulated in data portion **1010** of TCP message **1006**. Interworking message **1012** includes a header portion **1014** for indicating the message type and a data portion **1016** containing the actual

message. If the header portion **1014** indicates that the interworking message is a multipart message, data portion **1016** of interworking message **1012** includes a first field **1018** indicating the number of messages present in the multipart message. After the first field, the multipart message can include one or more interworking messages **1012A** to **1012N**.

DTMF Digit Handling

Another extension of the base protocol provided by the agent interworking protocol is dual tone multifrequency (DTMF) digit handling. Numerous studies have concluded that encoding and transporting DTMF digits within the media stream is not suitable for supporting networked services, such as credit card validation, automated services, and voice mail, which require DTMF digit recognition. Some of these services require recognition not only of the tone being transmitted but also of the duration of the tone. Algorithms for encoding voice and data can distort the tone and/or change the duration of the tone sought to be transmitted. As a result, the receiving application might not be able to correctly interpret the tone.

Currently, a fully standardized approach for handling the transport of DTMF digits after call establishment is not defined. The approach that is currently in the most favor is to encode DTMF digits as a special real time protocol (RTP) payload type that is exchanged between devices participating in the media stream. If this approach is standardized, various digital signal processor (DSP) manufacturers must comply with the standard before interworking will be accomplished. Because this involves a hardware change, much time can elapse before this occurs.

As a current solution to the problem, some of the agent protocols have implemented out-of-band techniques for handling DTMF digits. The agent interworking protocol according to embodiments of the present invention preferably provides a mapping to and from the out-of-band DTMF digit handling techniques of supported protocols. In order to provide a method for communicating DTMF information between protocol agents, the agent interworking protocol defines a data structure referred to herein as the digit information parameter. FIG. **11** illustrates an exemplary digit information parameter data structure. In the illustrated embodiment, the digit information parameter data structure includes a digit field **1100** and a duration field **1102**. Digit field **1100** is capable of storing a digit value indicative of the DTMF digit being transmitted. For example, the digit field can contain a numerical value that indicates one of the keys on a telephone handset. Duration field **1102** stores a duration value for indicating the duration of the tone represented by the digit in the digit field. Specific examples of when the digit information parameter is exchanged will be explained below with reference to the call flow diagrams.

Interworking Examples

A. H.323 Fast Start to MGCP

FIG. **12** is a call flow diagram illustrating exemplary call signaling performed by H.323 and MGCP agents including interworking agent capabilities according to an embodiment of the present invention. In FIG. **12**, H.323 endpoint **1200** is seeking to establish a call with an MGCP endpoint through MGCP gateway 1202. MGCP gateway **1202** performs both signaling gateway and media gateway functions. H.323 agent **1204** includes interworking agent functionality and implements an originating call model. MGCP agent **1206** includes interworking agent functionality and implements a terminating call model.

In line **1** of the call flow diagram, H.323 endpoint **1200** sends a SETUP message to H.323 agent **1204**. The SETUP message includes fast start parameters that specify suggested media options for the initial media stream. In line **2** of the call flow diagram H.323 agent **1204** sends an agent interworking protocol initial address message (IAM) to MGCP agent **1206**.

The AIP IAM message contains the media capabilities definition mapped from the fast start parameters extracted from the SETUP message. In line **3** of the call flow diagram, MGCP agent **1206** sends an MGCP create connection (CRCX) message to MGCP gateway **1202**. The CRCX message contains local connection options that are mapped from the AIP media capabilities information extracted from the IAM message into MGCP format. In line **4** of the call flow diagram, MGCP gateway **1202** sends an OK message to MGCP agent **1206**. The OK message includes a media capability selected by MGCP gateway **1202** from the media capabilities specified in the CRCX message. The media description for the selected capability is returned in the SDP portion of the OK message.

In line **5** of the call flow diagram, MGCP agent **1206** sends an AIP call progress (CGP) message to H.323 agent **1204**. An AIP CPG message is used to signal events other than release and answer between protocol agents implementing different call halves. In the illustrated example, the CPG message includes a mapping of the SDP portion of the OK message into AIP format. In addition, any other media capabilities which the MGCP agent is capable of supporting can be included in the media description. In line **6** of the call flow diagram, H.323 agent **1204** transmits an ALERTING message to H.323 endpoint **1200**. H.323 agent **1204** maps the media description from the AIP CPG message into fast start parameters and includes the fast start parameters in the ALERTING message. Any additional capabilities that were received by the H.323 agent are stored for later usage.

In line **7** of the call flow diagram, when the MGCP end user answers the call, signaling gateway **1202** sends a NOTIFY message to MGCP agent **1206**. The NOTIFY message alerts MGCP agent **1206** of the off-hook event. In line **8** of the call flow diagram, in response to the NOTIFY message, MGCP agent **1206** transmits an AIP answer message (ANM) to H.323 agent **1204**.

In line **9** of the call flow diagram, in response to the answer message, H.323 agent **1204** transmits a CONNECT message to H.323 endpoint **1200**. In lines **10** and **11** of the call flow diagram, H.323 endpoint **1200** and H.323 agent **1204** exchange master/slave and master/slave acknowledge messages. These messages are sent according to H.245 master/slave determination. This determination is made to resolve conflicts in media formats. H.323 agent **1204** handles the exchange and does not map the exchange to the agent interworking protocol.

In line **12** of the call flow diagram, H.323 endpoint **1200** transmits an H.245 terminal capabilities set (TCS) message to H.323 agent **1204** to communicate the media capabilities of endpoint **1200** to H.323 agent **1204**. In line **13**, H.323 agent **1204** acknowledges the TCS message. In line **14** of the call flow diagram, H.323 agent **1204** transmits a multipart message to MGCP agent **1206**. The multipart message includes the capabilities of the H.323 device mapped into AIP format. The capabilities are sent to MGCP agent **1206** in the AIP CPG message. Optionally, the H.245 representation of the TCS can be sent as well. In this case, a multipart

message is sent between call halves. The multipart message includes both the AIP CPG message and the H.245 TCS message. Because MGC 1206 might not support H.245 TCS, MGCP agent 1206 might, i.e., if H.245 TCS is not supported, discard the TCS portion of the multipart message and process only the AIP portion.

In lines 15 and 16 of the call flow diagram, H.323 endpoint 1200 and H.323 agent 1204 exchange TCS and TCS ACK messages. In this exchange, the capabilities of the other end, i.e., of MGCP gateway 1202, which were received and stored upon receipt of the CPG message in line 5 of the call flow diagram, are sent to H.323 endpoint 1200 as an H.245 terminal capability set.

H.323 Non-Fast Start to MGCP

While FIG. 12 illustrated H.323 to MGCP interworking for H.323 fast start procedures, FIG. 13 illustrates H.323 to MGCP interworking without fast start procedures. In other words, in FIG. 13, the H.323 media capabilities are not exchanged until after call establishment. The entities involved in communications in FIG. 13 are the same as those illustrated in FIG. 12. Thus, a description of these entities is not repeated herein.

Referring to FIG. 13, in line 1 of the call flow diagram, H.323 endpoint 1200 sends a SETUP message to H.323 agent 1204. The SETUP message does not include fast start parameters. In line 2 of the call flow diagram, H.323 agent 1204 transmits an AIP IAM message to MGCP agent 1206. The IAM message contains no media description. In other words, the connection information parameter is either not included or set to a null value. In line 3 of the call flow diagram, MGCP agent 1206 transmits a CRCX message to MGCP gateway 1202. The CRCX message can optionally contain a default set of media capabilities that do not reflect capabilities supported by the H.323 endpoint. A CRCX message example is as follows:

CRCX:

R: HD

L: Default media capabilities

M: Inactive (or receive only)

In the CRCX message example, the hd value in the R field instructs gateway 1202 to go off-hook. The value in the L field specifies local connection options, which indicate to gateway 1202 the media capabilities of H.323 endpoint 1200. In response to the CRCX message, in line 4 of the call flow diagram, MGCP gateway 1202 transmits an OK message to MGCP agent 1206. The OK message includes an SDP portion with the media description for the connection. An exemplary media description is as follows:

v=O

c=IP address

m=media description

In the exemplary media description set forth above, the IP address in the c=parameter is the IP address on MGCP gateway 1202 for receiving the media stream. The media description specified in the m=parameter includes the type of media that the media gateway is capable of receiving, e.g., voice, data, or video.

In line 5 of the call flow diagram, MGCP agent 1206 transmits an AIP call progress (CPG) message to H.323 agent 1204. The AIP CPG message includes the connection information parameter data structure in which the media description from the SDP portion of the AIP message is mapped into AIP format. The media description is stored by H.323 agent 1204, but is not transmitted to H.323 endpoint 1200 as a fast start parameter.

In line 6 of the call flow diagram, H.323 agent 1204 transmits an ALERTING message to H.323 endpoint 1200. The ALERTING message notifies H.323 endpoint 1200 that the MGCP end user is being alerted. When the MGCP end user answers the call, in line 7 of the call flow diagram, MGCP gateway 1202 transmits a NOTIFY message to MGCP agent 1206. In line 8 of the call flow diagram, MGCP agent 1206 transmits an AIP answer message (ANM) to H.323 agent 1204. In line 9 of the call flow diagram, H.323 agent 1204 transmits a CONNECT message to H.323 endpoint 1200. In lines 10 and 11 of the call flow diagram, the H.245 master/slave determination takes place. H.323 agent 1204 handles the exchange and does not map the exchange to the agent interworking protocol.

In lines 12 and 13 of the call flow diagram H.323 endpoint 1200 and H.323 agent 1204 exchange H.245 TCS and H.245 TCS ACK messages. This exchange communicates the media capabilities of H.323 endpoint 1200 to H.323 agent 1204. In line 14 of the call flow diagram, H.323 agent 1204 transmits a multipart message to MGCP agent 1206. H.323 agent 1204 maps the capabilities of H.323 endpoint 1200 into AIP format and includes these capabilities in an AIP CPG message. Optionally, the H.245 representation of the TCS can be sent as well. In this example, a multipart message is sent that includes both the AIP CPG message and the H.245 TCS message. Since MGCP agent 1206 can not support H.245 TCS, MGCP agent 1206 can only process the AIP portion of the multipart message.

In lines 15 and 16 of the call flow diagram, H.323 endpoint 1200 and H.323 agent 1204 exchange TCS and TCS ACK messages. In this exchange, the capabilities of MGCP gateway 1202, which were received and stored upon receipt of the AIP CPG message in line 5 of the call flow diagram, are sent to H.323 endpoint 1200 as an H.245 terminal capability set. In lines 17-20 of the call flow diagram, H.323 endpoint 1200 and H.323 agent 1204 exchange H.245 OPEN LOGICAL CHANNEL and H.245 OPEN LOGICAL CHANNEL ACKNOWLEDGE messages. In this exchange, H.323 agent 1204 two unidirectional media streams between the end users. H.323 endpoint 1200 returns its RTP port for each media stream in the acknowledge messages.

In line 21 of the call flow diagram, H.323 agent 1204 transmits an AIP CPG message to MGCP agent 1206. CPG message contains an updated media description to reflect the RTP address of H.323 endpoint 1200 and the mode change to send receive. In line 22 of the call flow diagram, MGCP agent 1206 transmits a modify connection message to MGCP gateway 1202. This updates the media description at MGCP gateway 1202 and a full duplex media path between end users is complete.

H.323—NAQ.931 Call Hold Scenario

FIG. 14 illustrates interworking between an H.323 endpoint and an NAQ.931 device for a call hold scenario. In FIG. 14, it is assumed that a bi-directional media stream has been established between H.323 endpoint 1200 and NAQ.931 device 1400. H.323 endpoint 1200 can be an IP terminal, as previously described with respect to FIG. 12. NAQ.931 device 1400 can comprise an IP terminal. H.323 agent 1402 includes interworking agent capabilities as well as H.323 gatekeeper capabilities. Similarly, NAQ.931 agent 1404 includes interworking agent capabilities as well as NAQ.931 agent capabilities.

In line 1 of the call flow diagram, NAQ.931 device 1400 transmits a HOLD message to NAQ.931 agent 1404. In line 2 of the call flow diagram, in response to the HOLD

message, NAQ.931 agent **1404** transmits an AIP CPG message to H.323 agent **1402**. The CPG message includes the connection information parameter data structure. The channel operation field in the data structure is set to mode change, and the mode field in the data structure is set to inactive. In line **3** of the call flow diagram, H.323 agent **1402** transmits a TCS=0 message to H.323 endpoint **1200**. In line **4** of the call flow diagram, H.323 agent **1402** transmits a CLOSE LOGICAL CHANNEL message to H.323 endpoint **1200**. The CLOSE LOGICAL CHANNEL message doses one of the two channels between H.323 endpoint **1200** and NAQ.931 device **1400**. In line **5** of the call flow diagram, NAQ.931 agent **1404** transmits a FACILITY message to NAQ.931 device **1400**. The FACILITY message indicates that inactive mode has been entered. In line **6** of the call flow diagram, H.323 endpoint **1200** transmits a CLOSE LOGICAL CHANNEL ACKNOWLEDGE message to H.323 agent **1402** acknowledging the closing of logical channel **1**.

In lines **7** and **8** of the call flow diagram, H.323 endpoint **1200** and H.323 agent **1402** exchange H.245 CLOSE LOGICAL CHANNEL and CLOSE LOGICAL CHANNEL ACKNOWLEDGE messages for logical channel **2**. Once logical channel **2** is closed, in line **9** of the call flow diagram H.323 agent **1402** transmits an AIP CPG message to NAQ.931 agent **1404**. The AIP CPG message includes the connection information parameter. The change operation field in the connection information parameter data structure is set to mode change, and the mode is set to inactive.

H.323—NAQ.931 Call Retrieve Scenario

FIG. **15** illustrates H.323 to NAQ.931 interworking for a call retrieve scenario. The entities illustrated in FIG. **15** are the same as those illustrated in FIG. **14**. Hence, a description thereof is not repeated herein. In FIG. **15**, it is assumed that a call between H.323 endpoint **1200** and NAQ.931 device **1400** has been put on hold. Thus, the signaling that must occur between H.323 endpoint **1200** and NAQ.931 device **1400** to retrieve the call includes reopening the logical channels between H.323 endpoint **1200** and NAQ.931 device **1400**.

In line **1** of the call flow diagram illustrated in FIG. **15**, NAQ.931 device **1400** transmits a RETRIEVE message to NAQ.931 agent **1404**. In line **2** of the call flow diagram, NAQ.931 agent **1404** transmits an AIP CPG message to H.323 agent **1402**. The CPG message includes the connection information parameter data structure with the change operation field set to mode change and the mode set to send/receive. In lines **3** and **4** of the call flow diagram, the H.245 master/slave determination occurs between H.323 endpoint **1200** and H.323 agent **1402**. H.323 endpoint **1200** and H.323 agent **1402** must revert to a TCS exchange in order to reestablish the media streams. Accordingly, in lines **5** and **6** of the call flow diagram H.323 endpoint **1200** and H.323 agent **1402** exchange TCS and TCS ACK messages.

In line **7** of the call flow diagram H.323 endpoint **1200** transmits an H.245 OPEN LOGICAL CHANNEL to H.323 agent **1402**. In line **8** of the call flow diagram, H.323 agent **1402** transmits an AIP CPG message to NAQ.931 agent **1404**. The CPG message includes the connection information parameter data structure. The channel operation field and the data structure is set to mode change and the mode is set to send only. This reestablishes one of the media streams between H.323 endpoint **1200** and NAQ.931 device **1400**. In line **9** of the call flow diagram, NAQ.931 agent **1404** transmits a FACILITY message to NAQ.931 device **1400**. The FACILITY message includes a mode parameter that sets the channel to be receive only.

In line **10** of the call flow diagram H.323 agent **1402** transmits an OPEN LOGICAL CHANNEL ACKNOWLEDGE message to H.323 endpoint **1200** acknowledging the opening of logical channel **1**. In line **11** of the call flow diagram, H.323 agent **1402** transmits an H.245 OPEN LOGICAL CHANNEL message to H.323 endpoint **1200** to open logical channel **2**. In line **12** of the call flow diagram, H.323 endpoint **1200** transmits an H.245 OPEN LOGICAL CHANNEL ACKNOWLEDGE message to H.323 agent **1402**. In line **13** of the call flow diagram, H.323 agent **1402** transmits an AIP CPC message to NAQ.931 agent **1404**. The CPG message includes the connection information parameter data structure. The channel operation field in the data structure is set to mode change and the mode field is set to send/receive. In line **14** of the call flow diagram, NAQ.931 agent **1404** transmits a FACILITY message to NAQ.931 device **1400**. The FACILITY message includes a mode field that sets the mode to be send/receive. Once this message is received, both logical channels are open between H.323 endpoint **1200** and NAQ.931 device **1400**.

H.323—MGCP Hold Scenario

FIG. **16** illustrates interworking for an H.323—MGCP hold scenario. In FIG. **16**, it is assumed that a call has been established between H.323 endpoint **1200** and MGCP device **1600**. MGCP device **1600** is assumed to support an event package for call hold and retrieve events. H.323 device **1200** is the same as H.323 device **1200** described with respect to FIG. **12**, and hence a description thereof is not repeated herein. MGCP device **1600** can be an MGCP device, such as a media gateway. MGCP agent **1602** includes interworking agent functionality as well as MGCP media gateway controller functionality. H.323 agent **1402** is the same as H.323 agent **1402** described with respect to FIG. **14**, and hence a description thereof is not repeated herein.

In line **1** of the call flow diagram, MGCP device **1600** transmits a NOTIFY message to MGCP agent **1602**. The NOTIFY message includes an event that informs MGCP agent **1602** that the end user connected to MGCP device **1600** has placed the call on hold. In line **2** of the call flow diagram, MGCP agent **1602** transmits an AIP CPG message to H.323 agent **1402**. The AIP CPG message includes the connection information parameter. The channel operation field in the connection information parameter data structure is set to mode change and the mode field is set to inactive. In line **3** of the call flow diagram, H.323 agent **1402** transmits a TCS=0 message to H.323 endpoint **1200**.

In lines **4** and **5** of the call flow diagram, H.323 endpoint **1200** and H.323 agent **1402** exchange CLOSE LOGICAL CHANNEL messages to close the logical channel between H.323 endpoint **1200** and MGCP device **1600**. In line **6** of the call flow diagram, MGCP agent **1602** transmits a MODIFY connection (MDCX) message to MGCP device **1600** indicating that the mode has been set to inactive. In lines **7** and **8** of the call flow diagram, H.323 endpoint **1200** and H.323 agent **1402** exchange the messaging required to close logical channel **2**. H.323 agent **1402** interprets the inactive mode as a hold and applies H.323 third party calls and rerouting procedures to implement the hold actions. These procedures result in the closing of both unidirectional media streams between H.323 endpoint **1200** and MGCP device **1600**. In line **9** of the call flow diagram, H.323 agent **1402** transmits an AIP CPG message to MGCP agent **1602**. The CPG message includes the connection information parameter data structure. The channel operation field in the connection information data structure is set to mode change, and the mode field is set to inactive.

H.323—MGCP Retrieve Scenario

FIG. 17 illustrates an H.323—MGCP retrieve scenario. The entities illustrated in FIG. 17 are the same as those illustrated in FIG. 16, and hence a description thereof is not repeated herein. In FIG. 17, it is assumed that a call between H.323 endpoint 1200 and MGCP device 1600 has been placed on hold.

In line 1 of the call flow diagram, MGCP device 1600 transmits a NOTIFY message to MGCP agent 1602. The NOTIFY message contains an event that indicates to MGCP agent 1602 that the end user connected to MGCP device 1600 wishes to retrieve the call. In line 2 of the call flow diagram, MGCP agent 1602 transmits an AIP CPG message to H.323 agent 1402. The AIP CPG message includes the connection information parameter data structure. The channel operation field in the data structure is set to mode change, and the mode field is set to send/receive.

In lines 3 and 4 of the call flow diagram, H.323 agent 1402 and H.323 endpoint 1200 make a master/slave determination. The H.323 devices must revert to a TCS exchange in order to reestablish the media streams. Accordingly, in lines 5 and 6 of the call flow diagram, H.323 agent 1402 and H.323 endpoint 1200 exchange terminal capabilities.

In line 7 of the call flow diagram, H.323 endpoint 1200 transmits an H.245 open logical channel message to H.323 agent 1402 to open one of the logical channels between H.323 endpoint 1200 and MGCP device 1600. In line 8 of the call flow diagram, H.323 agent 1402 transmits an AIP CPG message to MGCP agent 1602. The AIP CPG message includes the connection information parameter data structure. The change operation field in the data structure is set to mode change, and the mode field is set to send only. In line 9 of the call flow diagram, MGCP agent 1602 transmits a modify connection message to MGCP device 1600. The modify connection message includes a mode field setting the mode to receive only. In line 10 of the call flow diagram H.323 agent 1402 acknowledges the H.245 open logical channel messages transmitted in line 7 of the call flow diagram.

In lines 11 and 12 of the call flow diagram, H.323 endpoint 1200 and H.323 agent 1402 exchange messaging for opening the other logical channel between H.323 endpoint and MGCP device 1600. In line 13 of the call flow diagram, H.323 agent 1402 transmits an AIP CPG message to MGCP agent 1602. The CPG message includes the connection information parameter data structure. The channel operation field in the data structure is set to mode change, and the mode field is set to send/receive. In line 14 of the call flow diagram, MGCP agent 1602 transmits a modify connection message to MGCP device 1600. The modify connection message instructs the device to change the mode to send/receive. At this point, both media streams between H.323 endpoint 1200 and MGCP device 1600 are established.

H.323 to MGCP Primary Rate Interface (PRI)

Common Channel Signaling (CCS)

FIG. 18 illustrates exemplary call signaling for H.323 to MGCP PRI (CCS). In FIG. 18, signaling gateway 1800 relays call control signaling between a circuit-switched network and a packet-switched network. For example, signaling gateway 1800 may be connected to a PSTN end office on one side and to an IP network on the other side. In the illustrated embodiment, signaling gateway 1800 is configured to forward Q.931 call signaling messages from the PSTN network to a packet network and vice-versa. On the circuit-switched side, SG 1800 can be configured to send and receive Q.931 over Q.921 call signaling messages. On the packet-switched side, signaling gateway 1800 can be configured to send and receive Q.931 over ISDN User Adaptation over TCP/IP messages.

Media gateway 1802 converts between packets and circuits to communicate the media stream to and from a PSTN end user device. On the circuit-switched side, media gateway 1802 can send and receive the media stream using pulse code modulation (PCM) voice. On the packet-switched side, media gateway 1802 can send and receive the media stream using RTP over UDP/IP. In the illustrated embodiment, media gateway 1802 is controlled using MGCP.

CCS agent 1804 exchanges call control information with signaling gateway 1800 and media control information with media gateway 1802. In the illustrated embodiment, CCS 1804 communicates with signaling gateway 1800 using Q.931 call signaling over IUA over TCP/IP and with media gateway 1802 using MGCP. CCS agent 1804 also communicates with H.323 agent 1806 using the agent interworking protocol, as described above. It is understood that CCS agent 1804 and H.323 agent 1806 can be part of a call server. H.323 agent 1806 and H.323 gateway 1808 exchange call signaling information according to ITU Recommendation H.225.

In line 1 of the call flow diagram, signaling gateway 1800 transmits a SETUP message to CCS agent 1804. The SETUP message includes information, such as the dialed digits for creating a call with the called party. In line 2 of the call flow diagram CCS agent 1804 transmits a CALL PROCEEDING message to signaling gateway 1800 to indicate that CCS agent 1804 is attempting to establish a call with the called party. In line 3 of the call flow diagram, CCS agent 1804 sends an MGCP CREATE CONNECTION message to media gateway 1802. In response to the CREATE CONNECTION message, media gateway 1802 transmits an ACKNOWLEDGE message including an SDP portion that specifies the supported media capabilities of media gateway 1802. In line 5 of the call flow diagram, CCS agent 1804 transmits an AIP IAM message to H.323 agent 1806. The AIP IAM message includes the connection information parameter that specifies the supported media capabilities of media gateway 1802. In line 6 of the call flow diagram, H.323 agent 1806 transmits a SETUP message specifying the media capabilities of media gateway 1802 to H.323 to H.323 gateway 1808. In line 7 of the call flow diagram, H.323 gateway agent 1808 transmits an ALERT message to H.323 agent 1806 indicating that the called part is being alerted, of the incoming call. The ALERT message includes the supported media capabilities of H.323 gateway 1808. In line 8 of the call flow diagram, H.323 agent 1806 sends an AIP CALL PROGRESS message including the connection information parameter specifying the media description of H.323 gateway 1808. In line 9 of the call flow diagram, CCS agent 1804 transmits an ALERT message to signaling gateway 1800 indicating that the called party is being alerted.

In line 10 of the call flow diagram, CCS agent 1804 transmits an MGCP MODIFY CONNECTION message specifying the mode of the connection as receive only and including thc media description of H.323 gateway 1808. In line 11 of the call flow diagram, media gateway 1802 acknowledges the MODIFY CONNECTION message.

In line 12 of the call flow diagram, when the called party answers the call, H.323 gateway 1808 transmits a CONNECT message to H.323 agent 1806. In line 13 of the call

flow diagram, in response to receiving the CONNECT message, H.323 agent **1806** transmits an AIP ANSWER message to CCS agent **1804**. In line **14** of the call flow diagram, CCS agent **1804** transmits a CONNECT message to signaling gateway **1800** indicating that the call has been answered. In line **15** of the call flow diagram, CCS agent **1804** transmits a MODIFY CONNECTION message to media gateway **1802** opening the connection as send/receive. In line **16** of the call flow diagram, media gateway **1802** acknowledges the MODIFY CONNECTION message. At this point, a bi-directional media stream communication is established between the called and calling parties. Thus, the call flow diagram illustrated in FIG. **18** embodies the true MGCP reference architecture whereby the SG and MG are separate entities.

MGCP—H.323 Call Setup and Exchange of DTMF Digits

FIG. **19** illustrates call signaling and exchange of DTMF digits between an MGCP gateway and an H.323 gateway. The entities illustrated in FIG. **19** are the same as those illustrated in FIG. **16**. Hence, a description thereof is not repeated herein.

In FIG. **19** it is assumed that a connection has already been established between H.323 endpoint **1200** and MGCP device **1600**. Therefore, call setup and teardown messages are not shown.

In line **1** of the call flow diagram, H.323 endpoint **1200** transmits a user input indication message to H.323 agent **1402** that includes the DTMF digit * encoded in the message. In line **2** of the call flow diagram, H.323 agent **1906** transmits an AIP_INFO message to MGCP agent **1602** that indicates that information is being communicated to MGCP agent **1602**. The AIP_INFO message includes the digit information parameter that specifies the DTMF digit entered by the end user connected to H.323 gateway **1902**. In line **3** of the call flow diagram, MGCP agent **1602** transmits a MODIFY CONNECTION message to MGCP device **1600**. The modify connection message includes a signal indicating the DTMF digit *. In line **4** of the call flow diagram, MGCP device **1600** acknowledges the modify connection message. The remaining messages in FIG. **19** are similar to the messages described with respect to lines **1–4** of the call flow diagram. Hence, a description thereof is not repeated herein.

It will be understood that various details of the invention can be changed without departing from the scope of the invention. Furthermore, the foregoing description is for the purpose of illustration only, and not for the purpose of limitation—the invention being defined by the claims.

What is claimed is:

1. A call server comprising:

(a) a fist protocol agent for communicating with a first internet protocol (IP) telephony device according to a first IP telephony protocol;

(b) a second protocol agent for communicating with a second IP telephony device according to a second IP telephony protocol; and

(c) an interworking agent for providing functions usable by the first and second protocol agents to communicate with each other according to a third protocol, the functions provided by the third protocol being a superset of functions provided by the first and second IP telephony protocols, said interworking agent further adapted to determine that a first parameter associated with the first IP telephony protocol does not map to the second IP telephony protocol and communicating first parameter to the second protocol agent without alteration.

2. The call server of claim **1** wherein the interworking agent comprises a first interworking agent component associated with the first protocol agent and a second interworking agent component associated with the second protocol agent.

3. The call server of claim **1** wherein the first protocol agent is a media gateway control protocol (MGCP) agent, the first IP telephony protocol is MGCP, the second protocol agent is an International Telecommunications Union (ITU) Recommendation H.323 agent, and the second IP telephony protocol is H.323.

4. The call server of claim **1** wherein the first protocol agent is an International Telecommunications Union Recommendation H.323 agent, the first IP telephony protocol is **11323**, the second protocol agent is a session initiation protocol (SIP) agent, and the second IP telephony is SIP.

5. The call server of claim **1** wherein the first protocol agent is an International Telecommunications Union Recommendation H.323 agent, the first IP telephony protocol is H.323, the second protocol agent is a Bellcore Q.931 agent, and the second IP telephony protocol is an extension of Bellcore Q.931.

6. The call server of claim **1** wherein the first protocol agent is a media gateway control protocol (MGCP) agent, the first IP telephony protocol is MGCP, the second protocol agent is a media gateway control protocol (MGCP) agent, and the second IP telephony protocol is MGCP.

7. The call server of claim **1** wherein the first protocol agent is an International Telecommunications Union Recommendation H.323 agent, the first IP telephony protocol is H.323, the second protocol agent is an H.323 agent, and the second IP telephony protocol is H.323.

8. The call server of claim **1** wherein the first protocol agent performs originating call half functions and the second protocol agent performs terminating call half functions.

9. The call server of claim **1** wherein the interworking agent is adapted to provide a connection information parameter data structure usable by the first and second protocol agents, for communicating media capabilities and media stream management information between the first and second protocol agents.

10. The call server of claim **1** wherein the interworking agent is adapted to provide a digit information parameter usable by the first and second protocol agents for communicating dual tone multifrequency (DTMF) digits between the first and second protocol agents.

11. A method for interworking devices that communicate using different internet protocol (IP) telephony protocols, the method comprising:

(a) receiving, from a first telephony device, a first message formatted according to a first IP telephony protocol;

(b) in response to receiving the first message, generating a second message, formatted according to a second protocol, said second protocol being distinct from said first protocol, the second message including at least one of a media capabilities description and media stream management information derived from the first message;

(c) transmitting the second message to a second protocol agent; and

(d) in response to receiving the second message, generating a third message formatted according to a third IP telcphony protocol, the third message including at least one of the media capabilities description and media stream management information derived from the second message.

12. The method of claim 11 wherein receiving a first message includes receiving the first message formatted according to the media gateway control protocol (MGCP) and generating a third message includes generating the third message formatted according to ITU Recommendation H.323.

13. The method of claim 11 wherein receiving a first message includes receiving the first message formatted according to the session initiation protocol (SIP) and generating a third message includes generating the third message formatted according to ITU Recommendation H.323.

14. The method of claim 11 wherein receiving a first message includes receiving the first message formatted according to ITU Recommendation H.323 and generating a third message includes generating the third message formatted according to Bellcore Q.931.

15. The method of claim 11 wherein receiving a first message includes receiving the first message formatted according to ITU Recommendation H.323 and generating a third message comprises generating the third message formatted according to media gateway control protocol (MGCP).

16. The method of claim 15 wherein receiving the first message formulated to ITU Recommendation H.323 includes receiving the first message containing H.323 fast start parameters, wherein generating a second message include mapping the H.323 fast start parameters to a media capabilities description in the second message, and generating the third message includes mapping the media capabilities description to MGCP.

17. The method of claim 11 wherein receiving a first message includes receiving a HOLD message from the first telephony device, generating the second message includes generating a message including a connection information parameter having a mode change value for changing the mode of a media stream communication associated with the first telephony device, and wherein generating a third message includes generating a message for changing the mode of the media stream communication to inactive according to the third IP telephony protocol.

18. The method of claim 11 wherein receiving a first message includes receiving a RETRIEVE message from the first telephony device, and generating a second message includes generating a message including a connection information parameter having a mode change value of active.

19. The method of claim 11 wherein receiving a first message includes receiving a first message including at least one dual tone multifrequency (DTMF) digit value, generating a second message includes mapping the DTMF digit value to a digit information parameter value in the second protocol, and generating a third message includes mapping the digit information parameter value to a DTMF digit value formatted according to the third IP telephony protocol.

20. The method of claim 11 comprising transmitting the third message to a second telephony device configured to communicate according to the third IP telephony protocol.

21. A method for tunneling messages between protocol agents, the method comprising:

(a) receiving, from a first telephony device, a first message formatted according to a first IP telephony protocol;

(b) determining whether a parameter in the first message maps to a second IP telephony protocol;

(c) in response to determining that the parameter in the first message maps to the second IP telephony protocol, formulating a second message formatted according to the second IP telephony protocol; and

(d) in response to determining that the parameter in the first message does not map to the second IP telephony protocol, transmitting the first message without alteration to a second protocol agent.

22. A method for tunneling messages between protocol agents, the method comprising:

receiving, from a first telephony device, a first message formatted according to a first IP telephony protocol;

determining whether a parameter in the first message maps to a second IP telephony protocol;

in response to determining that the parameter in the first message maps to the second IP telephony protocol, formulating a second message formatted according to the second IP telephony protocol;

in response to determining that the parameter in the first message does not map to the second IP telephony protocol, transmitting the first message without alteration to a second protocol agent, and

in response to determining that the parameter in the first message partially maps to the second IP telephony protocol, formulating a multiprotocol message, the multiprotocol message including a message formatted according to the first IP telephony protocol and a third message formatted to the second IP telephony protocol.

23. The method of claim 22 comprising transmitting the multiprotocol message to a second protocol agent.

24. The method of claim 23 comprising in response to receiving the multiprotocol message, dividing the multiprotocol message into the second and third messages.

25. The method of claim 24 comprising after dividing the multiprotocol message, determining whether processing of the second message is supported by the second IP telephony protocol agent, and in response to determining that the processing of the second message is supported, processing the second message.

26. The method of claim 25 comprising processing the third message.

27. A computer program product comprising computer-executable instructions embodied in a computer readable medium for performing steps comprising:

invoking a first protocol agent for communicating with a first internet protocol (IP) telephony device according to a first IP telephony protocol;

invoking a second protocol agent for communicating with a second IP telephony device according to a second IP telephony protocol;

mapping media capabilities information extracted from messages received from the first and second IP telephony devices formatted according to the first and second IP telephony protocols to a third protocol;

transmitting a first message containing the media capabilities information and formatted according to the third protocol between the first and second protocol agents;

determining whether a parameter from the first IP telephony protocol maps to the second IP telephony protocol; and

in response to said determining that the parameter from the first IP telephony protocol does not map to the second IP telephony protocol, transmitting the parameter without alteration to the second protocol agent.

28. The computer program product of claim 27 wherein invoking a first protocol agent includes invoking a first protocol agent for performing originating call functions and invoking a second protocol agent includes invoking a second protocol agent for performing terminating call functions.

29. The computer program product of claim 27 comprising at the first protocol agent, mapping media stream information rceeived from the second protocol agent to the first IP telephony protocol.

30. The computer program product of claim 27 comprising, at the second protocol agent, mapping media stream information received from the first protocol agent to the second IP telephony protocol.

31. The computer program product of claim 27 wherein the first IP telephony protocol is the media gateway control protocol and the second IP telephony protocol is ITU Recommendation H.323.

32. The computer program product of claim 27 wherein the first IP telephony protocol is ITU Recommendation H.323 and the second IP telephony protocol is Bellcore Q.931.

33. The computer program product of claim 27 wherein the first IP telephony protocol is the session initiation protocol and the second IP telephony protocol is ITU Recommendation H.323.

34. A computer readable medium having software stored thereon, said software comprising;

a fist protocol agent for communicating with a first internet protocol (IP) telephony device according to a first IP telephony protocol;

a second protocol agent for communicating with a second internet protocol telephony device according to a second IP telephony protocol, wherein said second IP telephony protocol is distinct from said first IP telephony protocol;

a third protocol agent for communicating with a third internet protocol telephony device according to a third IP telephony protocol, wherein said third IP telephony protocol is distinct from said first and second IP telephony protocols; and

an interworking protocol adapted to represent a partial superset of messaging capabilities of said first, second, and third IP telephony protocols such that messages received in any of said first, second, or third IP telephony protocols from a first IP device are converted to said interworking protocol and then translated into a different one of said first, second, or third IP telephony protocols for transmission to a second IP device.

* * * * *